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24

The D-MAC/packet System for Satellite and Cable

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Contents

	Page
Introduction *by B. Salkeld*	3
Satellite Broadcasting in the United Kingdom *by M.D. Windram, G. J. Tonge and R.C. Hills*	5
An Overview of the D-MAC/packet System	11
The Vision Signal	16
Packet Construction and Transmission	24
Sound Channels	30
Interpretation Blocks	38
Service Identification Signals	41
Data Broadcasting	48
High Priority Information – Line 625	50
The Conditional Access System	55
RF Transmission of D-MAC/packet	63

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In compiling the description of the D-MAC/packet system, reference has been made to the Specification of the systems of the MAC/packet family Tech. 3258E published by the European Broadcasting Union. This specification should be consulted for further interpretation of the D-MAC/packet system.

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Introduction

by B. F. Salkeld
Head of Satellite Engineering
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The UK Government first asked the IBA to report on the prospects for launching a viable DBS project in 1985. The IBA was able to report that, despite the uncertainties, there appeared to be sufficient market interest to justify a go-ahead. The Government authorised the IBA to proceed, and a three-channel 15-year contract was advertised on 2nd April 1986. The contract on offer was for appointment as a programme contractor who would be responsible for the space segment. The broadcast uplink station would be provided by the IBA, as the broadcaster.

A programme contract specification set out the requirements for IBA clearance of programme schedules, a proper proportion of advertising time, and limitations to ensure that control of the programme company did not lie outside the EEC. A spacecraft specification set out the minimum acceptable performance of the spacecraft in terms of its coverage, channel filtering and spacecraft antenna requirements.

The specification was aimed to achieve certain minimum requirements while allowing flexibility in the design to encourage innovation.

The coverage was specified in terms of a minimum EIRP of 59 dBW to be directed towards defined boundary locations. Both the minimum coverage zone and the power level were smaller than that permitted under the World Broadcasting and Satellite Administrative Radio Conference (WARC77) plan and were intended to give the opportunity for design flexibility with the aim of achieving a commercially attractive solution. The power was judged to be sufficiently low so as to allow a modest spacecraft size, yet being sufficiently higher than that of the proposed FSS services to offer distinct competitive advantage in the ease of reception. Eclipse protection was not made mandatory but the attention of applicants was drawn to the need to consider whether operation during eclipse might represent an advantage in terms of programming or as regards satellite reliability. Applicants had to satisfy the IBA on the degree of redundancy to protect consumer interests and were required to present viable launch proposals.

Other aspects of the specification included requirements for the spacecraft power, reaction and thermal control systems; these mostly required the applicant company to provide sufficient information to demonstrate that its proposition was realistic.

In December 1986 the IBA announced the contract award to British Satellite Broadcasting Limited (BSB). The group had guaranteed initial capitalisation of £200 million with a second round of funding to follow once the satellite had been launched. Programming proposals were for four distinct services over the three channels.

Intense negotiation for the satellite contract then took place until May 1987 when the Hughes Aircraft Company were named as the preferred supplier.

For the IBA contract award the various contenders had to demonstrate a viable launch proposition. Negotiations took place at a particularly unfortunate time concerning launch and insurance availability.

Negotiations for the satellite contract became increasingly competitive and it was possible eventually to arrange 'provision in orbit' through the chosen supplier, which would itself make the launch and insurance arrangements. The Hughes 376 satellite is to be carried on a Delta 4925 launcher in August 1989, with the second launch on a Delta 6925 taking place less than a year later.

The WARC77 Plan assumed FM transmissions and stipulated that any alternative system must not create worse interference. The IBA Specification called for a Multiplexed Analogue Component (MAC) coding system employing a 20 Mbit/s data burst in addition to the vision signal, and the D-MAC/packet system was chosen. No operating satellite has yet used D- MAC/packet and the IBA

will carry out a series of D-MAC/packet tests on the spacecraft payload before launch.

In addition, as an aid to receiver manufacturers, D-MAC/packet test transmissions from a satellite simulator mounted on the Croydon television mast in London will also be conducted prior to the launch date.

The business plan for UK-DBS is based on income from a combination of advertising and subscription. The D-MAC/packet standard allows for the effective scrambling of pictures, with a conditional access system to address individual decoders.

The main IBA broadcast uplink station is to be located on a site near Southampton. There is to be a pair of automatically steered 8 metre diameter uplink antennas driven from klystron amplifiers delivering approximately 1kW each into the combining unit. Three programme chains each have full one-for-one protection and the station will be run unattended from the IBA control centre in London. Component video signals and analogue sound are to be delivered by a British Telecom optical fibre link from the programme playout centre in London. These enter the D-MAC coders along with teletext and subscriber management information data. A compressed, scrambled and addressed D-MAC version of the input signal is used to frequency modulate a 70 MHz carrier feeding the radio frequency equipment.

As well as the main radio frequency equipment delivering 17 GHz broadcast uplink signals, the station is equipped with a comprehensive automated test system to prove ground station equipment and spacecraft payload.

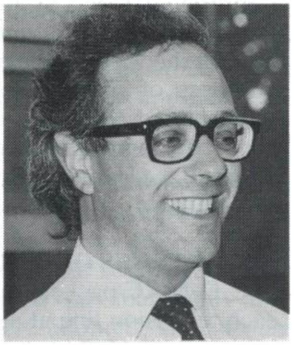
In autumn 1989 the world's first privately financed direct broadcasting satellite service will bring three new channels to the United Kingdom, within just four years of the Government having first asked the IBA to report on the prospects of a viable DBS project.

Not only will the service be a 'first', but also the method of transmission will be new, the D-MAC/packet system. The system, largely developed by the IBA, will allow broadcasters to provide signals that can be adapted for the needs of this decade and into the next century. This edition of the *IBA Technical Review* outlines the basis of the D-MAC/packet transmission system.

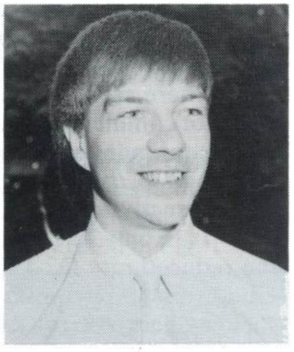
The IBA has good reason to be proud of its achievements in bringing the UK-DBS project to fruition and in developing the Multiplexed Analogue Component system.

Satellite Broadcasting in the United Kingdom

by M. D. Windram, G. J. Tonge, R. C. Hills



DR M. D. WINDRAM graduated in 1966 and then spent a further three years at Cambridge University on research in the field of radio astronomy. This was followed by two years in industry before joining the IBA Experimental and Development Department in 1971, where he was involved in the development of adaptive antenna systems and tunable equipments. He became Head of Radio Frequency Section in 1978 and then in 1982 Head of Video and Colour Section where he was involved with the development of the MAC video system. In 1987, he was appointed Head of the Experimental & Development Department, responsible for a wide range of broadcast research projects including studies on the future evolution of the MAC system, and on high-definition television.



DR G. J. TONGE joined the IBA in 1980 after completing postgraduate research in Applied Mathematics at Southampton University. He was involved in the image processing activities and HDTV work of the Authority's Experimental and Development Department until he took up his present post as Head of Engineering Secretariat in 1987.

R. C. HILLS joined the BBC after graduation from the University of Bristol in 1954, and remained until 1967 when he was appointed Head of Masts and Aerials Section of the IBA (then ITA). In 1969 he became Head of Station Design and Construction Department, and was responsible for the staff engaged on major capital works programmes to create the IBA's network of UHF television transmitters. He was appointed Chief Engineer (Transmitters) in 1973 and Assistant Director of Engineering (Operations) in 1978. He took up his present post as Assistant Director of Engineering (Corporate Development) in 1986 in which position he has responsibility for the IBA engineering work in support of the UK-DBS project. For the past ten years he has had special responsibility for the provision of consultancy services to overseas clients.

Synopsis
The United Kingdom is about to enter a new and exciting phase of its broadcasting history. In August 1989 the satellite will be launched for the world's first high-power DBS service which is funded entirely privately. For the commercial success of this service the broadcaster needs to establish a large market quickly and yet needs to offer the potential of receiver improvements to keep market interest. Against this background the D-MAC/packet system has been adopted as the signal format for the service. This system offers a reception option involving a relatively inexpensive set-top converter and hence should enable rapid market penetration. However, D-MAC also provides a number of service options which should stimulate ongoing market interest. For the vision part of the system four phases are described for the receiver: the existing receiver with set-top converter, the integrated MAC receiver, the wide-screen receiver and the HDTV receiver. The digital data part of the signal offers a variety of high-quality sound options as well as data services which represent an area of growing interest. They offer further opportunities for generating revenue for the broadcaster and in the context of HDTV offer a control channel by which improved receiver decoding can be effected.

Working backwards from the receivers and the transmission system, studios and sources are required which will feed existing and new services. In the UK environment the ideal HDTV studio standard will use 1250 lines and a 50-Hz field rate. This will interface well with the existing video and film environment and is ideally suited as a source for a high-definition MAC coding system for HDTV transmission.

INTRODUCTION

The IBA has chosen the D-MAC/packet system as the technical standard for the DBS service. This system belongs to the EBU MAC/packet family of standards[1]. This approach has been adopted to meet the dual requirements of:

a. opportunities for diverse services and for future evolution;

b. the necessity for rapid service growth in the early stages.

The D-MAC/packet system has been chosen in preference to PAL and in preference to D2-MAC/packet in order to meet the first requirement, and in preference to 1125/60 HDTV in order to meet the second requirement.

ADVANTAGES OF MAC/PACKET SYSTEMS OVER PAL OR SECAM

The advantages of using the MAC system have been presented many times, but those features which enable future developments (requirement a. above) are, in summary:

The Use of Component Video rather than Composite Video

This gives a freedom from cross-colour and cross-luminance effects and hence an improved picture quality. The 'clean' signal that results can also be processed more readily for picture enhancement. Furthermore, the time-multiplexed approach of MAC offers better signal-to-noise performance than composite systems for an FM channel.

Digital Sound and Data

The MAC/packet system is a combined carrier of digital data and analogue vision. The data resource enables the provision of several high-quality sound channels as well as numerous data services.

Bandwidth Flexibility

The absence of subcarriers (for either colour information or for sound) in the MAC approach gives a flexibility to the vision bandwidth specification. One simple way of improving picture quality, for example, is to widen the bandwidth of the signal.

Dual Aspect Ratio Specification

The flexibility of the MAC/packet system is such that a picture aspect ratio of either 4:3 or 16:9 can be provided. This is particularly important with a future evolution to HDTV in mind, as will be discussed later.

Conditional Access System

It is becoming increasingly important to be able to control the access to television services, both for the purposes of offering subscription or pay-per-view services and also for copyright reasons. The MAC/packet system offers a very secure and flexible conditional access facility using over-air addressing of receivers.

ADVANTAGES OF D-MAC OVER D2-MAC

The D-MAC/packet system carries digital data on each television line at an instantaneous rate of 20.25 Mbit/s, while the rate with D2-MAC is precisely half of this, 10.125 Mbit/s. The primary advantage of D-MAC over D2-MAC for UK satellite broadcasting is therefore that it carries twice the data capacity. This can be expressed in many ways, for example:

- 8 high-quality sound channels instead of 4;
- 16 commentary sound channels instead of 8;
- greatly increased revenue-earning potential for teletext or data services.

It is this latter factor which is seen by many to be the overriding consideration in an environment in which there are increasing opportunities for the provision of commercial teletext or data services.

The optimum base-bandwidth for the data part of the D-MAC/packet signal is approximately 8.4 MHz, exactly the same as is specified for the vision part. It is only in a narrow-band environment that the reduced rate D2-MAC system needs to be used. Under these circumstances the reduction in data capacity has to be tolerated along with the inevitable reduction in resolution for the vision. The EBU specification[1] allows for simple transcoding between the two systems where required.

ADVANTAGES OF D-MAC OVER AN 1125/60 SYSTEM

If the requirement for a rapid service growth is considered, it can be seen why the D-MAC system has been adopted in favour of the 'new system/new market' approach as proposed for HDTV in Japan, where the 1125- line/60-Hz system is envisaged.

It is important to appreciate the financial climate for satellite broadcasting in the United Kingdom. DBS is a completely new service which in no way duplicates any of the existing terrestrial channels and, in addition, is privately funded with no financial support from the UK Government. The satellites are bought and owned by British Satellite Broadcasting (BSB). Receiver provision is a straightforward commercial venture on the part of the manufacturers.

There is, therefore, a climate in which rapid growth of service penetration is essential to provide the income needed by the broadcaster. For this, a rapid take-up of DBS receivers is essential and hence the receiver price (the cost of entry into the DBS service) is a very critical parameter.

D-MAC offers a relatively inexpensive option for entry into the DBS service by purchasing a receiving dish along with a set-top converter to feed an existing receiver. The expected cost of this equipment is of the order of £250. This is only marginally more than if the PAL system were adopted, and substantially less (by a tenfold factor) than the cost of entry into the service if a 60-Hz HDTV transmission system were used. This is because 60-Hz HDTV would require not only a receiving dish installation and set-top converter but also a completely new HDTV display. It should be noted that in Japan a relatively simple adapter for NTSC reception of HDTV DBS services has been developed, acknowledging the importance of rapid service penetration. Such an adapter for PAL reception in the United Kingdom would be out of the question, because of the different field-rates employed.

FOUR PHASES IN RECEIVER DEVELOPMENT

Four phases in the evolution of the satellite broadcasting services are envisaged which are typified by the four classes of receiver illustrated in Fig. 1.

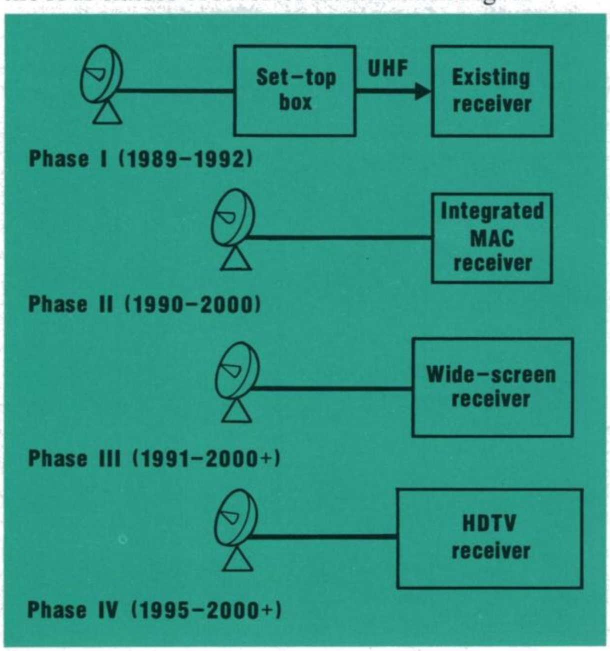


Fig. 1. Four phases in DBS receiver development.

Phase I represents the inexpensive entry cost option referred to above. The satellite receiving dish and outdoor unit feed the indoor unit (or set-top converter box) which, for most receivers in the United Kingdom, would plug into the conventional UHF antenna socket. In this case, the vision signal format is therefore converted from MAC into PAL and many of the advantages of using the MAC system appear to have been lost under these circumstances. This need not be the case for existing receivers equipped with a peritelevision connector since a component RGB output could also be available from the set-top converter box. Nevertheless, whatever the capability of existing receivers, it should be stressed that this phase is seen as essential for the starting up of a UK-DBS service and without it the subsequent phases would never happen. The sound signal will be available with the UHF television signal or, alternatively, as high-quality stereo to feed a hi-fi system or a modern television set equipped for this. This phase is expected to last from the beginning of the service (1989) for perhaps three years.

It is expected that Phase II will start very soon after the beginning of the service. This phase is typified by the integrated MAC receiver which has the 'set-top converter' built in. In this phase the benefits of the component format of the MAC signal will be available as well as the high-quality sound options. The anticipated cost of such a receiver is likely to be around £400- £500 for the lower end of the product range.

It is in Phase III, which may start as early as 1991, that another important vision characteristic of the D-MAC system is reflected in the receiver. This characteristic is the wide aspect ratio transmission format. A market is foreseen for 16:9 aspect ratio wide-screen receivers which give high-quality component video and stereo sound performance, but which do not have HDTV processing. From the start of the service in 1989, the IBA is expecting to transmit some programme material in wide-screen (16:9) format and by 1991 the proportion of material in wide-screen format will have increased. An important feature of the D-MAC decoders used in Phases I and II is that they will include a second expansion factor so that either 4:3 or 16:9 broadcasts can be displayed correctly on a conventional 4:3 receiver display[2]. Phase III receivers could be divided into two classes. The first (wide-MAC receivers) use the conventional 625/50/2:1 scanning format, while the second (EDTV receivers) incorporate scan up-conversion processing and use a double-rate scan for

either 100-Hz field rate or progressive scanning. An anticipated price for a wide-MAC receiver early in Phase III is of the order of £1,000, although this would be expected to fall as 16:9 display tubes are manufactured in larger quantities.

Phases I, II and III are seen as important preliminary steps prior to the HDTV phase - Phase IV. At some stage during Phase III some productions will become available in HDTV format and it is in Phase IV that the full picture-quality potential associated with this can be conveyed to the viewer. Whether the picture-quality improvement associated with HDTV, over and above that of wide- MAC or EDTV, is worthwhile will then be judged by the market.

HDTV services are provided by the use of HDTV sources (with a scanning standard of 1250/50/1:1, 1250/50/2:1 or 625/50/1:1, for example) combined with an HD-MAC transmission system. Such a system has an HDTV input and output but a MAC-compatible 625/50/2:1 transmission signal in the middle. The complexity of processing for HD-MAC could be similar to that of the 1125/60/2:1 MUSE transmission system, for a comparable picture quality. Alternatively, many believe that by using extensive motion-compensation techniques an HD-MAC quality, which is higher than that of MUSE, can be provided at the cost of extra complexity. It is unlikely that the HDTV receiver cost will be less than £1,500 at the estimated start of Phase IV in 1995.

This phased evolution towards HDTV is seen as preferable to a sudden step to a new HDTV system. As already discussed, a cheap option for entry into the DBS service (Phase I) is seen as essential for the financial success of DBS in the United Kingdom. Furthermore, HDTV is unlikely to be of significant domestic interest until a new display technology is available. This is seen as possible more on a 1995/2000 timescale, whereas the ripe time for the commencement of a new DBS service in the United Kingdom is earlier (1989). With this background the D-MAC/packet system offers the optimum solution for UK-DBS with its HDTV potential coupled with its inexpensive receiver option.

THE HDTV STUDIO

The possibility of future HDTV services has been introduced and so the HDTV studio standard is clearly of importance here. In the context of UK broadcasting, any HDTV studio/source must feed existing and new networks. A DBS studio complex of the future may be configured as illustrated schematically in Fig. 2.

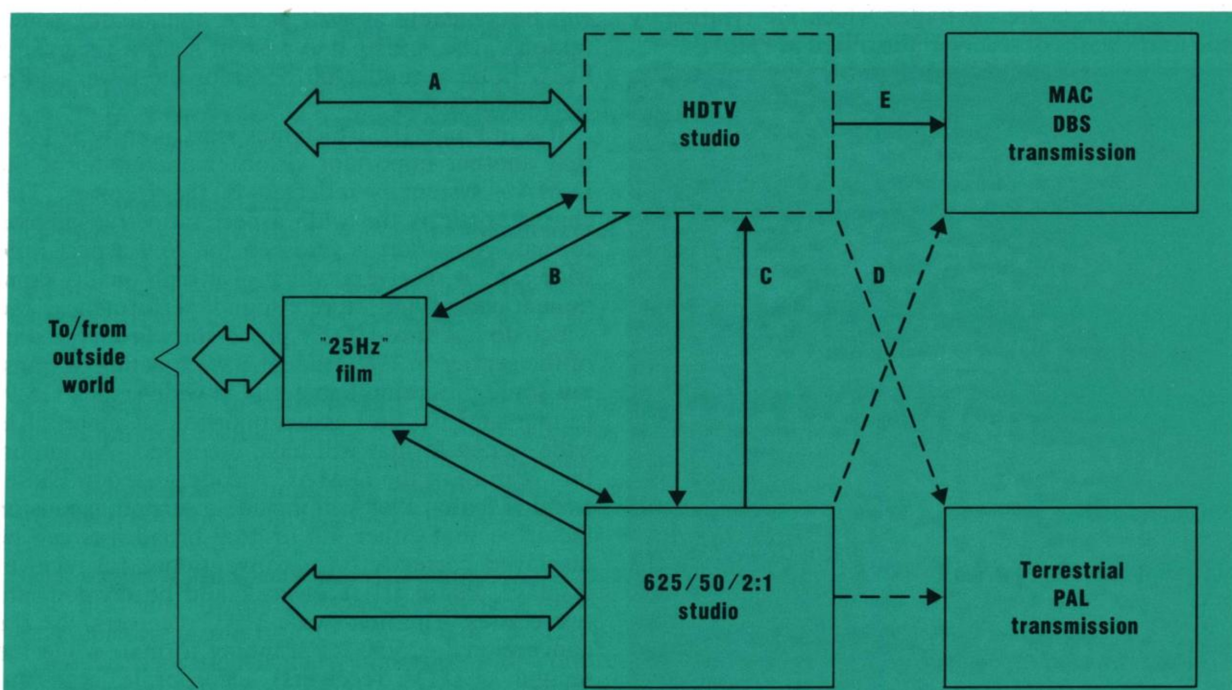


Fig.2. DBS studio of the future in the United Kingdom.

Five classes of interface with the HDTV studio standard are identified:

a. Interface with a (possibly different) HDTV standard used in other parts of the world.
b. Interface with motion-picture film with a frame rate of 24 Hz (or 25 Hz as used in conjunction with current UK television). As a source, film is likely to form a significant proportion of HDTV broadcasts, especially in the early stages of an HDTV service. In addition, electronic HDTV production is likely to be used increasingly in the making of feature films.
c. Interface with the 625/50/2:1 studio. As an input to the HDTV studio, archive material stored on the 625-line standard will be required from time to time to be included in HDTV productions. In the other direction, many HDTV productions will also be down-converted to a 625-line studio format, in order to be included in 625-line productions.
d. There may be a need (e.g. prestige sporting events) to broadcast the same programme content both in HDTV and over the terrestrial PAL network.
e. It is seen as essential that HDTV broadcasts are compatible with the 625/50/2:1 MAC transmission system.

Some comments relating to the benefits/deficits connected with these interfaces for either a 50-Hz or a 60-Hz HDTV studio are collected together in Table 1.

Table 1

SUMMARY OF BENEFITS/DEFICITS OF A 50 or 60-Hz HDTV STUDIO FOR THE UNITED KINGDOM STUDIO OF FIG. 2

Interface	50-Hz HDTV Studio	60-Hz HDTV Studio
A	HDTV field-rate conversion may be necessary for exchanged material.	HDTV field-rate conversion may not be necessary for exchanged material.
B	Simple high-quality conversion to and from 25-Hz film.	More complex lower-quality conversion to and from 24-Hz film.
C,D	Simple conversion with high quality.	Complex conversion with motion artifacts.
E	Compatible MAC transmission format possible.	HDTV field-rate conversion need for all programme material prior to compatible MAC transmission.

When considering the comments in Table 1 it soon becomes clear that there is only one interface (A) for which there may be some advantage for the UK broadcaster in adopting a 60-Hz HDTV system - assuming that other parts of the world adopt 60 Hz. For all the other interfaces, a 50-Hz HDTV studio standard provides both a more economic and a substantially higher-quality solution. In particular, a standard of 1250/50/1:1 (with initial equipment operating on 1250/50/2:1 or 625/50/1:1) is the ideal for studios related to UK broadcasting.

This conclusion has clear implications for the debate concerning a single worldwide standard for HDTV studios. There are two possible solutions:

- adopt a 50-Hz standard worldwide; this would then give problems of conversion in the 60-Hz countries;
- adopt HDTV studio standards as higher members of the extensible family of digital standards of CCIR Recommendation 601 (i.e. 1250/50 and 1050/59.94).

While the second of these solutions does not strictly provide a 'single worldwide standard', it offers several useful features. These include ease of commonality of HDTV equipment, simpler compatible broadcasting options in an NTSC environment and a fuller establishment of the already agreed Recommendation 601 standard. In realistic terms, it may offer the only compromise solution which maintains order in a field which may otherwise result in a proliferation of unrelated standards.

CONCLUSIONS

The choice of the D-MAC/packet system for the UK-DBS service has been linked to the dual requirements of:

- opportunities of diverse services and for future evolution (e.g. HDTV);
- the necessity for rapid service growth in the early stages.

This second requirement has been emphasised strongly in the light of the financial background to the UK service.

An evolution towards HDTV receivers through four phases has been identified and the suitability of a 1250/50 HDTV studio standard has been emphasised in this context.

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An Overview of the D-MAC/ packet System

Synopsis
An appreciation of the overall concept of the D-MAC transmission standard is given in this chapter. The nature of Muliplexed Analogue Components is covered leading into a description of the vision waveform, the reasons for time compression and the chosen sampling rates. Before the particulars of the sound and data channels are considered, attention is also given to the choice of duobinary coding and the formation of the packet structure. The basis of scrambling and encryption techniques is mentioned followed by the cable requirements for D and D2-MAC.

It is hoped that having read this chapter the reader will be able to appreciate the content of any of the individual chapters and not necessarily need to consider them in a strict order.

INTRODUCTION

The D-MAC/packet standard is an entirely new broadcast system providing a highly flexible medium for transmission of vision, sound and data. It forms part of the EBU family of MAC/packet systems and has been adopted by the United Kingdom for Direct Broadcasting by Satellite (DBS). The flexible structure is designed to adapt to the needs of future services. From its initial conception, MAC has been tailored for transmission by satellite, making efficient use of the 27 MHz DBS channel bandwidth yet without displacing the use of existing television receivers.

The essence of the system is the transmission of a baseband time division multiplex in which a time-compressed analogue picture signal is combined with a 20.25 Mbit/s duobinary data burst which carries sound, data and digital synchronisation signals. A considerable degree of compatibility with existing European television receivers is achieved by maintaining the same scanning standard of 625 lines, 50 Hz field rate and 2:1 interlace.

Compatibility will also be maintained when future transmissions contain pictures with an aspect ratio of 16:9, wider than the present day 4:3 format. (It will be possible to convey pan control information allowing existing 4:3 aspect ratio receivers to display the appropriate part of the wide-screen picture). Further development leading to higher definition pictures can also be accommodated within D-MAC. For this a form of Digitally Assisted Television (DATV) is envisaged in which control data is carried in the vertical blanking interval, enabling a special receiver to reconstruct a 1250-line display.

A 10 microsecond duobinary burst placed at the beginning of each line provides a high overall useful data rate of approximately 3 Mbit/s. This gives great flexibility, allowing as many as 8 high quality (15 kHz bandwidth) sound channels to be conveyed; alternative formats include a combination of sound channels, data and teletext services, together with over-air addressing for conditional access. The digital sound and data is organised into packets. These are blocks of information related to particular services, easily allowing the number and type of sound and data services to be varied to suit the need of the broadcaster.

In addition to the above use of the burst, the content of the multiplex is signalled to the receiver by a Service Identification system which makes use of data packets with address '0' and by information contained in line 625. Line 625, made up entirely from digital data, also organises the structure of the complete sound/data and vision multiplex.

VISION

Multiplexed Analogue Components (MAC)

The vision signal is made up of lines of time-compressed colour-difference and luminance components. These are formed from component analogue (or digital) picture lines which have been digitally sampled and time-compressed, so that the colour-difference and the luminance components can be accommodated in the standard active line period of approximately 52 microseconds. Each line of vision is preceded by a 206-bit duobinary data burst forming a complete line of 64 microseconds, as illustrated in Fig.1.

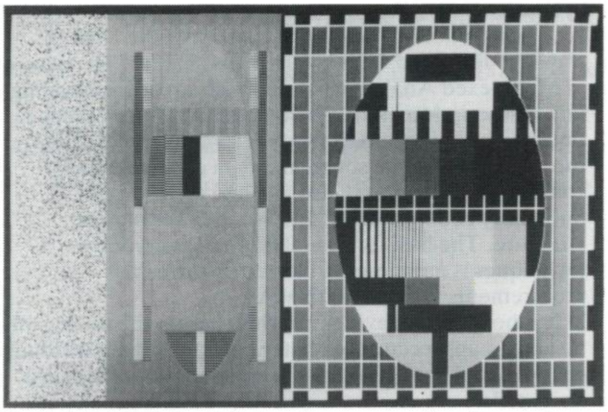


Fig.1. A D-MAC signal displayed on a conventional monochrome monitor with the addition of conventional sync. pulses.

In the encoding process the colour-difference signals have a compression factor of 3:1 whilst the luminance signal is time compressed by a factor of 3:2. The uncompressed vision signals have frequency response characteristics which conform to CCIR Recommendation 601, that is, the colour-difference bandwidth is 2.75 MHz and the luminance bandwidth 5.75 MHz.

A consequence of time compression is to increase the bandwidth in direct proportion to the compression ratio; as a result the overall video bandwidth is about 8.5 MHz.

Time compression at source and the necessary expansion within the receiver are achieved by digitally sampling the analogue signal, storing the samples and reading them out at the appropriate rate. In order to accurately decode (and if necessary descramble) the MAC waveform, the chosen 20.25 MHz clock rate must be conveyed to the receiver. This is conveniently obtained from the 20.25 Mbit/s rate of the data burst carried on each line.

An important consideration in the choice of the effective sampling rate of 20.25 MHz was its relationship, through the two compression ratios, to the internationally agreed sampling frequencies for digital video studio signals which are 6.75 MHz for colour-difference and 13.5 MHz for luminance (CCIR Recommendation 601). In defining the waveform, therefore, it is convenient to divide the entire 64 microsecond line period into 1296 time slots, or samples, based on 49.4 ns. (1/20.25 MHz).

Once sampled and time compressed, the colour-difference and luminance components are placed in sequence on every active picture line. Figure 2 shows the vision components preceded by the sound/data burst.

Each colour-difference component is transmitted line sequentially with the $E'U_m$ component on the odd lines and the $E'V_m$ component on the even lines. In the receiver, reconstitution of the missing colour-difference components is achieved by a process of averaging adjacent lines. The delay caused by this process is allowed for by introducing a compensating one-line delay of the luminance component in the encoding process. Active picture lines are transmitted on lines 24 to 310 inclusive in the first field and lines 336 to 622 inclusive in the second field. Line 624 of each frame is allocated to reference signals, while the entire duration of line 625 is devoted to digital data. The field-blanking interval is reserved for possible future digital assistance to provide pictures with higher definition.

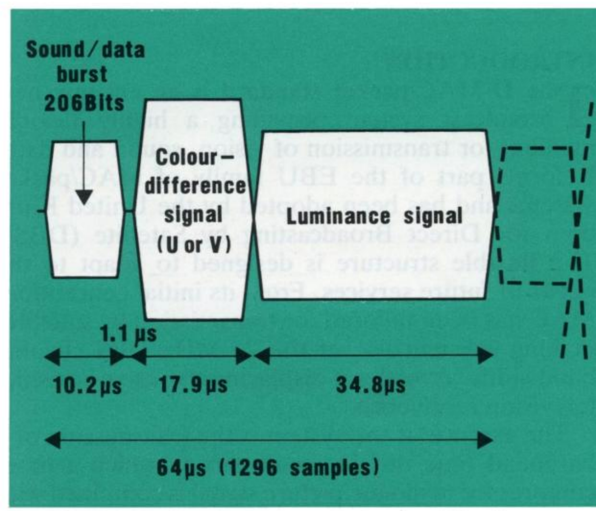


Fig.2. D-MAC active picture line with the sound/data burst.

Picture Synchronisation

All the frames of the D-MAC signal contain 625 lines at all times, thus eliminating any need for genlocking. Within each frame every line commences with a 6-bit word to provide for line and frame synchronisation. Frame synchronisation can also be obtained from a further sequence of bits transmitted within the data of line 625.

SOUND AND DATA

Data Coding

All the digital sound and data information in the multiplex is conveyed by a form of coding known as duobinary. With this coding system the signal has three characteristic levels rather than just two as with binary coding. A logic '1', in this case, is repre-

sented by an extreme 'positive' or 'negative' level, while the intermediate level represents a logic '0'. The use of duobinary offers the advantage over conventional binary coding in that it is more economical in terms of bandwith for a given data rate.

Packet Structure

As described above, the instantaneous bit rate of the digital information is 20.25 Mbit/s. A 206-bit burst is carried at the beginning of each line for the first 624 lines of each frame. (Line 625 consists entirely of 1296 bits of data). The majority of the burst, 198 bits per line, is used to convey the sound/data multiplex channel as shown in Fig.3.

The 198-bit burst on each line is divided into two sections of 99 bits which form, over 623 lines, into two subframes. Line by line, the bits in each subframe are arranged into discrete entities called packets where each packet contains 751 bits. Together, both subframes contain 164 packets (123,354 bits per television frame) which is equivalent to a total capacity of 3,083,850 bit/s.

The useful data contained in every packet is preceded by a header of 23 bits. The header allows the receiver to recognise and select the packets required for a particular service that is, sound, teletext or data while all other packets are rejected. A packet header also contains a continuity index which indicates to the receiver any packet loss (or gain) that may occur due to errors in the transmission path. In addition, all these packet 'overheads' are afforded a high degree of error protection by the application of redundancy coding techniques.

Sound

Within the high overall data capacity, the flexibility of the sound/data multiplex offers a number of possibilities for the transmission of sound channels together with additional data. Various combinations of sound channel types and data can co-exist at any

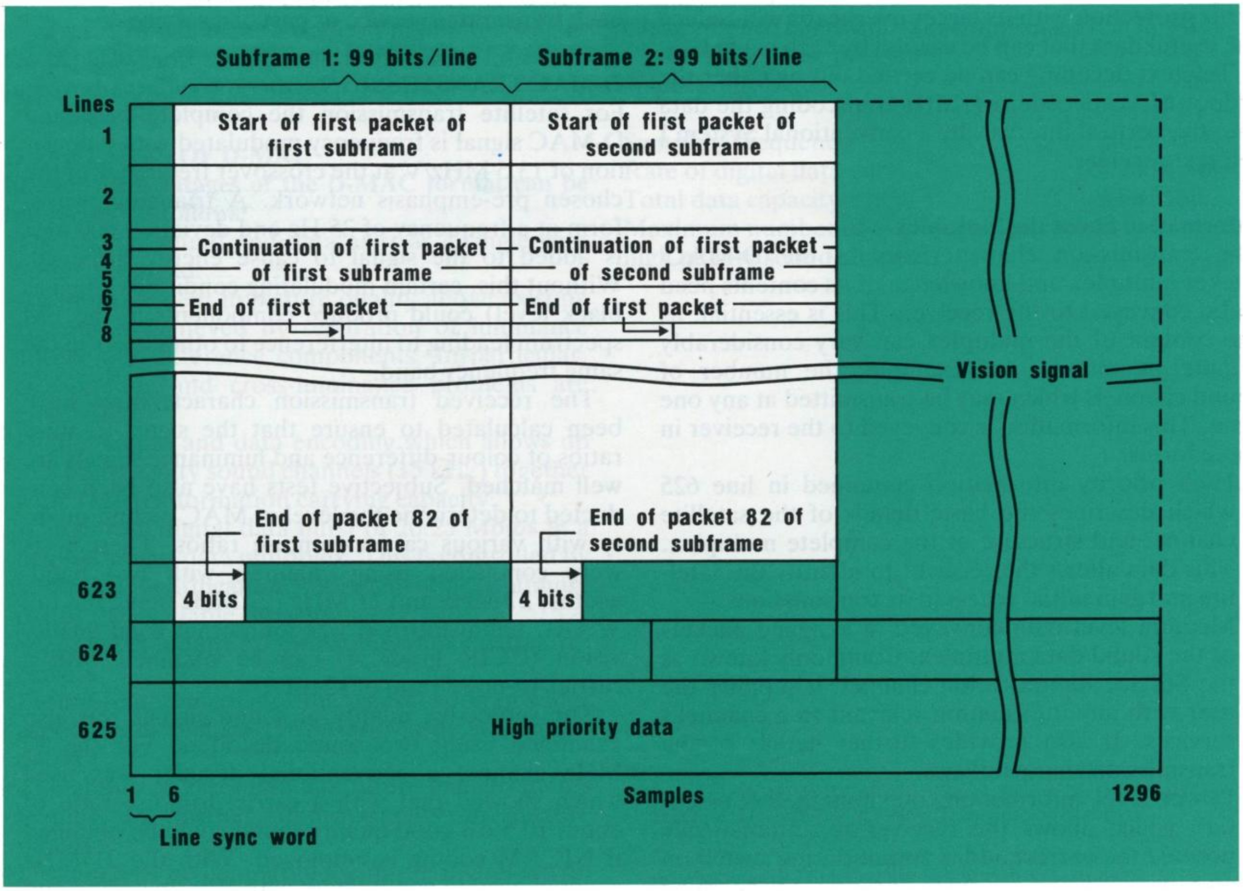


Fig.3. The packet multiplex (not to scale).

time, providing that in total they require less than the overall capacity of each sound/data subframe.

For high quality sound, a digital sampling rate of 32 kHz offers channels with an audio bandwidth of 15 kHz coded in either 14-bit linear or near- instantaneous companding 14-10 bit (NICAM) formats. The latter option is the same coding method to be used for the terrestrial UHF stereo sound transmissions in the United Kingdom. For commentary purposes, channels with medium quality sound of 7 kHz bandwidth at a digital sampling rate of 16 kHz can also be transmitted.

Teletext

The sound/data multiplex can also be used to carry teletext with a capacity far in excess of the present day terrestrial television services. In a similar manner to the sound channels, teletext is conveyed in packets which contain an address header followed by useful data. Two possible levels of error protection are provided. However, a choice of higher level error protection with its larger overheads will reduce the useful data that can be carried by each packet.

Teletext decoding can be carried out by either the indoor MAC decoder or, after transcoding the data into the vertical interval, by a conventional System I teletext receiver.

Information about the Multiplex

The transmission characteristics of the D-MAC/packet multiplex and knowledge of its contents need to be conveyed to the receiver. This is essential, as the content of the multiplex can vary considerably minute by minute; for example, the number of sound channels which may be transmitted at any one time. This information is conveyed to the receiver in three forms:

- High priority information contained in line 625 which describes the basic details of the satellite channel and structure of the complete multiplex. This data allows the receiver to identify the satellite and gain initial access to its transmissions.
- Medium level data conveyed in assigned packets of the sound/data multiplex. Commonly known as the Service Identification channel, it supplies the user with any information relevant to a channel's services. It also provides further details of the transmission characteristics.
- Lower level information, but nonetheless essential, which allows the receiver to automatically provide the correct audio output during switch-on and prepare the receiver for any forthcoming changes in the service. This data, known as Information Blocks, is placed in packets of the sound/data multiplex and is closely aligned with the relevant sound packets.

SCRAMBLING

Because the sound and the vision signals may be divided into discrete samples, the MAC waveform lends itself most easily to scrambling techniques. For the United Kingdom all forms of D-MAC transmission, including the sound and teletext, will be scrambled. The programme provider may, as he wishes, allow receivers to decode the signals free of charge. This can be achieved by sending the receiver the correct key signals to 'unlock' the scrambling. The process of locking the scrambled signals with hidden keys is known as encryption. Alternatively the programme provider can decide to send the unlocking keys only upon payment by the customer. A technique exists, developed by IBA engineers, whereby customers may be individually addressed over air for each transmitted service or part of a service.

D-MAC TRANSMISSION

For satellite transmission the complete baseband D-MAC signal is frequency modulated with a deviation of 13.5 MHz/V at the crossover frequency of the chosen pre-emphasis network. A triangular waveform at a frequency of 25 Hz and deviation 600 kHz is added to the signal to cause energy dispersal. Without this, certain modulating conditions (such as black level) could produce components in the FM spectrum leading to interference to other users in the same frequency band.

The received transmission characteristics have been calculated to ensure that the signal-to-noise ratios of colour-difference and luminance signals are well matched. Subjective tests have also been conducted to determine the level of MAC picture quality with various carrier-to-noise ratios. These tests were conducted using channels with two bandwidths, 27 MHz and 21 MHz (27 MHz being the full WARC bandwidth). It was found that good quality vision (CCIR grade 4) can be obtained with a carrier-to-noise ratio of 12 dB.

The subjective quality of sound signals was also examined using two sound decoders. For the 27 MHz channel a conventional decoder was used which showed that with a carrier-to-noise ratio of about 10.5 dB good quality sound could be obtained if NICAM coding is employed. With the 21 MHz channel used in conjunction with Viterbi coding and

linear Hamming coding, a carrier-to-noise ratio of 8 dB still allowed good sound quality.

CABLE

As the D-MAC signal requires a baseband bandwidth in the order of 8.5 MHz, it is well suited for transmission by modern cable installations. Because cable systems normally use Amplitude Modulation - Vestigial Side Band (AM-VSB), the frequency modulated satellite signals must be demodulated to baseband then remodulated to suit the cable transmission system. With a compressed vision bandwidth of about 8.5 MHz, the overall r.f. bandwidth is in the order of 10.5 MHz (this includes an allowance of 0.75 MHz vestige width to which is added a 1 MHz filter and guard band requirement). This is suitable for all cable networks with 12 MHz channel spacing. For those cable networks with a smaller channel spacing of perhaps 7 or 8 MHz, D-MAC signals can be translated to D2-MAC which, with a 10.125 Mbit/s data burst, has half the data capacity of D-MAC. However, this will incur a tradeoff in half the sound/data capacity and a reduction in picture quality because of the reduced bandwidth.

ADVANTAGES OF D-MAC

The many advantages of the D-MAC format can be summarised as follows:

- The vision, sound and data exist as a single-wire baseband signal
- An improvement in picture quality above existing systems is achieved by separation of luminance and colour-difference components. In particular, cross-colour and cross-luminance problems are eliminated.
- Digital sound and data encoding which allows up to 8 high quality sound channels (15 kHz) together with a large information carrying capacity.
- A high-rate digital data burst of 20.25 Mbit/s in a flexible sound/data multiplex. This is controlled by a service-identification system making optimum use of the available capacity.
- Scanning standard is compatible with existing receivers.
- Future requirements for the transmission of wider aspect ratio picture (16:9) are easily accommodated.
- The ability to accommodate future enhancements leading to further improvements in picture quality.
- Designed to be compatible with the digital studio standard.
- Scrambling for conditional access is easily accomplished.
- Simplified receiver design, with a single demodulator equally suitable for both vision and data.
- Can be broadcast by FM satellite transmission or AM-VSB cable distribution.

The following table summarises the essential characteristics of the D-MAC format.

Essential details of the D-MAC system

Carrier/noise ratio to produce picture quality better than grade 4	12 dB
Number lines per frame	625
Interlace	2:1
Horizontal frequency	15,625 kHz
Vertical frequency	50 Hz
Synchronisation	digital, in data burst
Luminance bandwidth (uncompressed)	5.75 MHz
Luminance compression	3:2
Chrominance bandwidth (uncompressed)	2.75 MHz
Chrominance compression	3:1
Aspect ratio	4:3 or 16:9
Sampling frequency	20.25 MHz
Rate of digital data burst	20.25 Mbit/s
Total data capacity	3.08 Mbit/s
Maximum number of high quality sound channels	8

References

1. K. Lucas & M. D. Windram. Direct Television Broadcasts by Satellite. Desirability of a New Transmission Standard. IBA E & D Report 116/81.
2. M. D. Windram, G. Tonge & R. Morcom. MAC – A Television System for High Quality Satellite Broadcasting. IBA E & D Report 118/82.

The Vision Signal

Synopsis
Within this chapter a comparison is made between the structure of D-MAC and PAL. The format of the baseband D-MAC waveform is described together with the composition of the complete 625-line frame. Reasons are given for the use of vision time compression and the consequences upon the bandwidth are discussed along with the sequential transmission format of the colour-difference signal. This chapter also includes sections devoted to synchronisation, clamping, vision scrambling and arrangements for the transmission of wider aspect ratio pictures.

INTRODUCTION

The basic scanning standard of the D-MAC *transmission* system is identical to that of the 625-line PAL system. For transmission the number of lines per frame is 625 of which 574 are active with the display based on a 2:1 interlace format of two fields at a 50 Hz rate. Each line has a duration of 64 microseconds and carries alternately one of the two colour-difference signals, $E'U_m$ and $E'V_m$. Synchronisation is derived digitally.

A requirement to transmit colour-difference and luminance information on each picture line may be accomplished by using either of two fundamental techniques, frequency or time division multiplexing. PAL uses the former where a subcarrier allows the two components to be frequency interleaved. In the case of MAC the latter technique of time division processing is adopted, whereby the duration of both the colour-difference and luminance signals is reduced, and each transmitted in sequence. Maintaining a time separation of the components ensures that major disadvantages of the PAL system are eliminated; that is, cross-colour and cross-luminance. The resulting improvement in picture quality compared to the PAL system is illustrated in Fig.1.

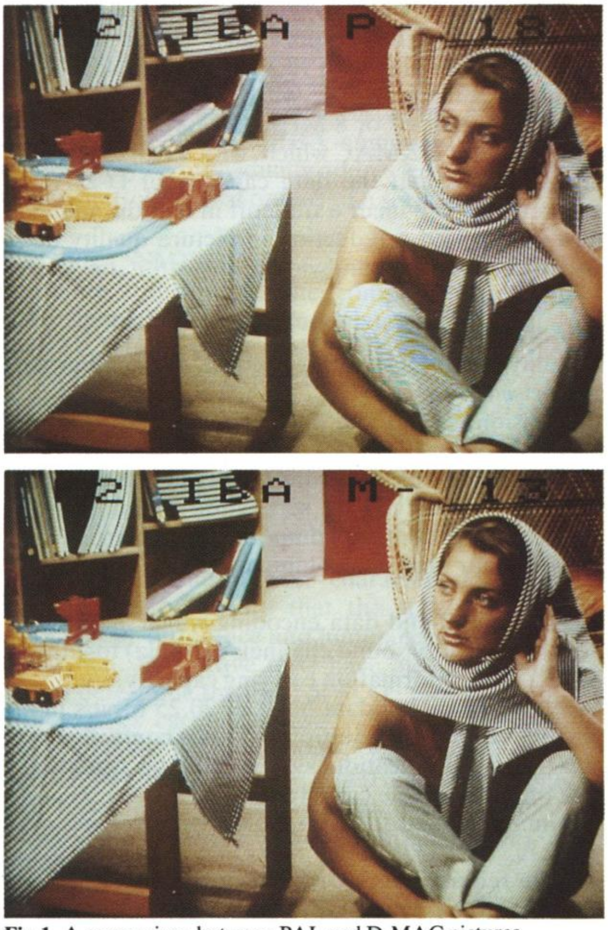


Fig.1. A comparison between PAL and D-MAC pictures.

Formation of the Picture Line

A time compression factor of 3:2 is applied, using digital techniques, to 697 samples representing the luminance component. This reduces the duration from about 52 microseconds to approximately 34.7 microseconds. Similarly each of the two colour-difference components, $E'U_m$ and $E'V_m$, is compressed by a factor of 3:1 to occupy a period of about 17.3 microseconds, 349 samples.

The components placed in sequence form a single picture line with an active period of 52 microseconds. Each line, therefore, contains one of the two colour-difference components together with the luminance component. Preceding the picture information is a 206-bit burst having a data rate of 20.25 Mbit/s. Thus, with the addition of clock and

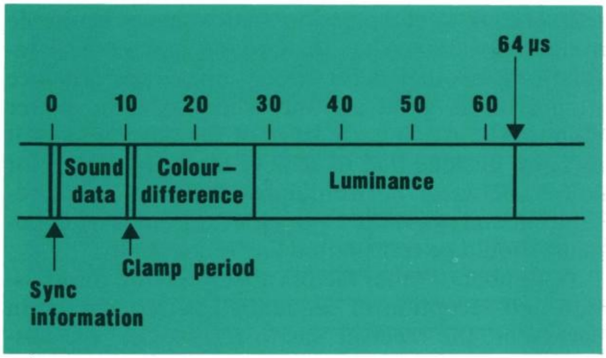


Fig.2. The time division multiplex.

transition periods, the total line duration extends to 64 microseconds. The overall result is a time-division multiplex of the form shown in Fig 2.

The Vision Waveform

As with PAL, the value of the MAC luminance voltage E'_Y is related to the colour-separation signal voltages E'_R, E'_G and E'_B by the standard equation ($E'_Y = 0.299\ E'_R + 0.587\ E'_G + 0.114\ E'_B$). From this relationship two colour-difference voltages are formed, E'_B-E'_Y and E'_R-E_Y and from which $E'U_m$ and $E'V_m$ are derived respectively. (The reader should note that for simplicity within the diagrams the colloquial terms U and V are used). These voltages, within the waveform, are scaled to have maximum amplitudes of 1 volt peak-to-peak and 1.3 volt peak-to-peak for the luminance and colour-difference signals respectively. Within these boundaries the luminance signal, which is transmitted on every line, can vary in content from -0.5 volt at black level to 0.5 volt peak-white. The colour-difference signals when at the maximum amplitude of 0.65 volt will correspond to 100% saturation, at 100% amplitude. All these levels are with reference to a zero-volt clamping level which immediately follows the data burst. The two colour-difference components are placed sequentially on alternate lines, odd lines of the frame carrying the $E'U_m$ component and even lines the $E'V_m$ component. A schematic baseband signal waveform is shown in Fig. 3.

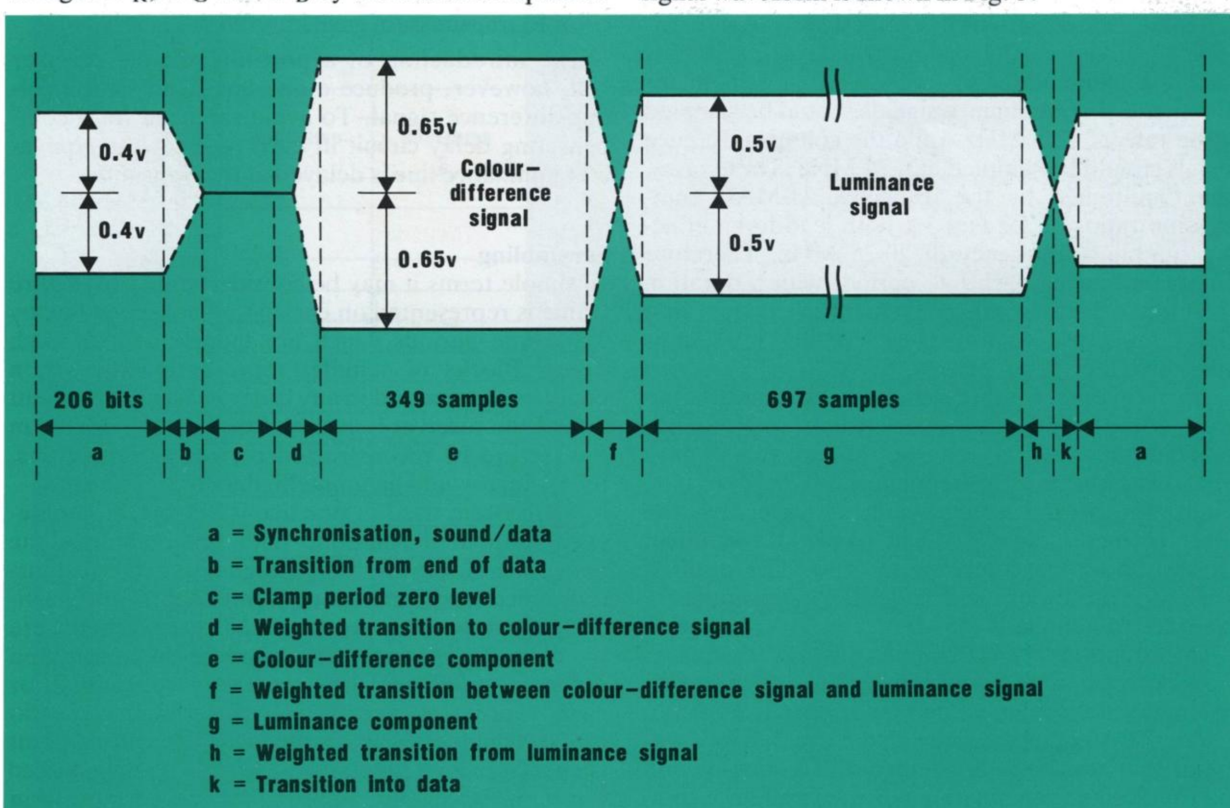


Fig.3. D-MAC/packet system – approximate baseband signal waveform for unscrambled pictures transmission. Clock frequency 20.25 MHz. (not to scale).

The Vision Frame

Two fields are formed by transmitting the picture information on lines 24 to 310 and 336 to 622 inclusive. Prior to each picture field, lines 23 and 335 provide a reference level by carrying a luminance signal held at - 0.5 volt (black level). A field-blanking interval exists between the two fields as shown in Fig. 4. The lines within the interval may be used for test purposes, additional sound/data or, in the future, additional duobinary data to enable enhancements to the picture quality.

At the bottom of each frame, line 624 contains amplitude reference information while the whole of line 625 is devoted to digital data relating the organisation of the entire multiplex.

Digital Processing

Each line of a complete waveform comprising data and video signals has a duration of 64 microseconds but it may also be considered as a number of discrete sample points; these are illustrated in Fig. 3. The sample points are a result of the specific design of the MAC standard which is based on the internationally agreed digital coding standard for studios (CCIR Recommendation 601). The agreement recommends that luminance signals should be sampled at the rate of 13.5 MHz while the colour-difference signals should be sampled at 6.75 MHz. These rates, when multiplied by the respective D-MAC compression ratios of 3:2 and 3:1 both lead to an effective sampling frequency of 20.25 MHz. Therefore when considering each line period, with a duration of 64 microseconds, it is convenient to divide it into 1296 sample slots or periods each of 49.4 ns, (that is 1/20.25 MHz.)

Prior to time compression, the baseband bandwidth of the luminance signal is from zero frequency to 5.75 MHz while the colour-difference signals have a bandwidth from zero frequency to 2.75 MHz. The application of time compression, however, has the effect of increasing the signal bandwidth in direct proportion to the compression ratio. The result is that at the output of the encoder the bandwidths are extended to nominally 8.5 MHz.

The adoption of these sampling frequencies, compression ratios and signal amplitudes was the result of lengthy mathematical and laboratory tests evaluated by IBA engineers. During the evaluation it was found that some needs conflicted. To comply with the CCIR Recommendation 601 the luminance component should have a sampling rate twice that of the colour-difference components, and it is desirable that the two compression ratios have a simple numerical relationship. Horizontal and vertical resolution together with video noise performance should be as good as, and where possible better than, PAL. At an early stage of the development it was also decided that to achieve the best balance for colour between horizontal resolution, vertical resolution and noise the two colour-difference components should be transmitted line sequentially.

A number of other factors arise from the transmission and reception of sequential colour signals. In particular, the receiver has to reconstitute the missing lines. This can be achieved by using a vertical 1,2,1 post-filter which takes the average of the adjacent two lines in the same field, in conjunction with a vertical 1,2,1 pre-filter prior to transmission. IBA tests revealed that this arrangement produces a good balance of the horizontal and the vertical resolution and yields good subjective results for most scenes. However, on some critical scenes, particularly certain moving captions, it was found that instead of 1,2,1 pre-filter the use of a 7 tap vertical pre-filter prior to transmission gave improved results.[1]

The introduction of a post-filter in the receiver will, however, produce a one line delay of the colour-difference signal. To avoid the need for a compensating delay circuit in each receiver the equivalent luminance line is delayed at the transmitter.

Scrambling

In simple terms it may be considered that the entire frame is represented on each 64 microsecond line by 1296 time periods which are independent of each other. Blocks of samples representing the vision could be transmitted in any order to the receiver, but only if the receiver is aware of the sending order can the picture be reconstructed correctly. Failing this, the picture would be unintelligible, or 'scrambled'.

The system used in practice is known as 'double-cut' component rotation. With this technique the active parts of the vision signal for the colour-difference and luminance components are partitioned into two segments each. These segments are then interchanged in time to produce the transmitted signal. The segment boundaries or 'cut points' for each component on each line are varied in a pseudo random manner down the picture. The cutting point is chosen from a 'flight-range' of 256 equally spaced positions which are approximately central within each component. Figure 5a. shows the flight-range to operate within the *active* part of the colour-

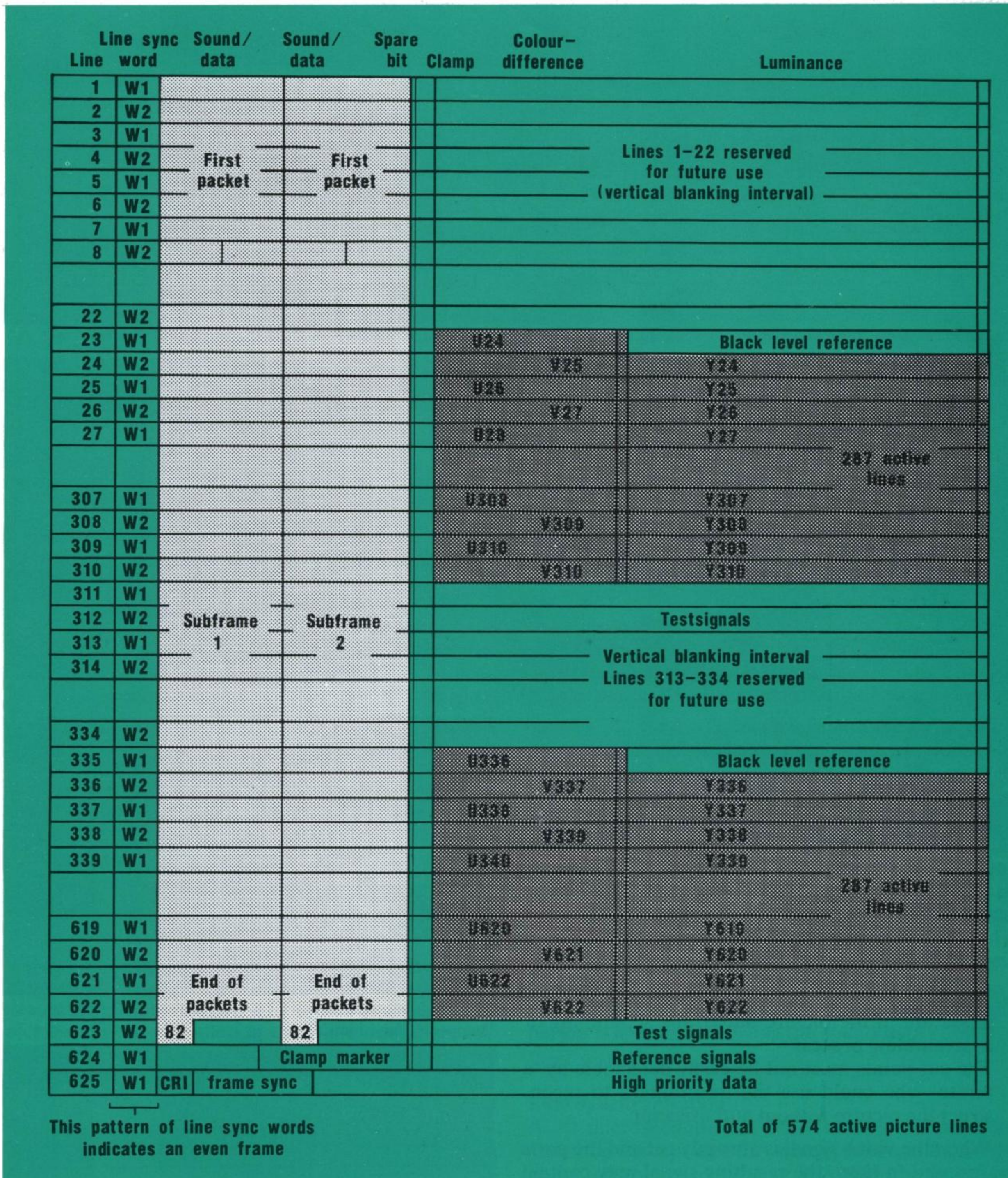


Fig.4. An even D-MAC/packet frame from the two frame sequence showing the alternating line sequence of the two colour-difference signals, each placed one line in advance of the corresponding line of luminance.

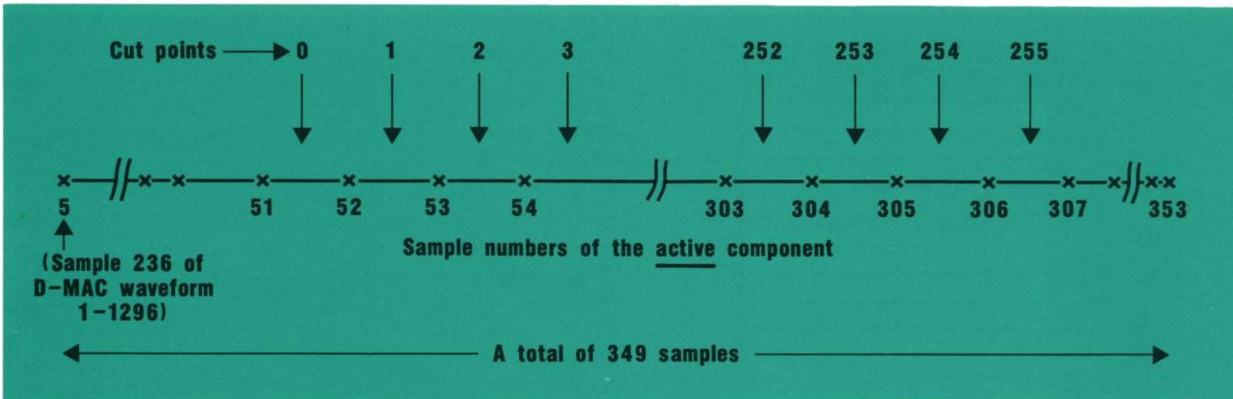


Fig.5a. Permissible cut-point positions for the colour-difference signal. The sample numbers relate to the *unscrambled* waveform.

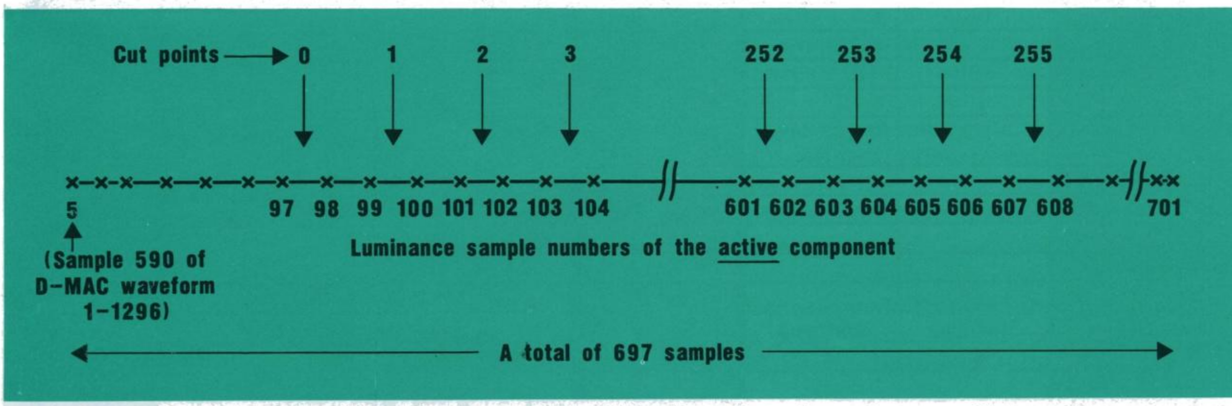


Fig.5b. Permissible cut-point positions for the luminance signal. The sample numbers relate to the *unscrambled* waveform.

difference component which has a total of 349 sample points. In a similar manner the 256 cutting points of the flight-range are centred within the 697 sample points of the *active* luminance component, as shown in Fig. 5b.

A receiver, when informed of the cut points, can reconstitute the segments to form the original signal as shown in a somewhat simplified manner in Fig. 6a and Fig. 6b.

The above process must be undetectable for two important, but different reasons:

- The viewer must not be able to detect any degradation of the picture due to the scrambling or descrambling process.
- The cut points must not be easily detectable by 'a pirate' who could use the information to reconstruct the picture without authorisation.

When the video signal is abruptly cut and the parts transposed in time, the resulting signal may contain edges or rapid transitions in level. These may occur at the cut points and at either end of the compo-

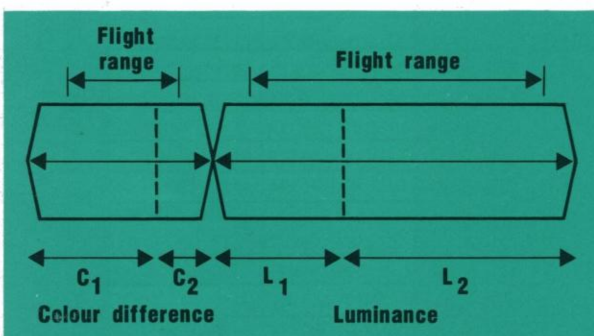


Fig.6a. Simplified diagram of non-scrambled waveform showing the possible cut-points.

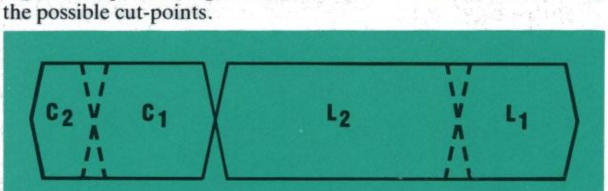


Fig.6b. Simplified diagram of waveform scrambled by double-cut component rotation.

nents. In conjunction with the effects that are caused by a non-perfect transmission path, visible defects could arise. These potential problems are overcome by symmetrical cross-fading of the waveform over three sample periods centred on the point where the join is to be made. This technique reduces the rise and fall times of the signal to within the normal system bandwidth. A representation of the scrambled waveform with cross-fading is shown in Fig 7.

For the above processing the analogue vision signal is digitally sampled within the encoder whose *output,* it should be noted, is a scrambled *analogue* signal.

alternative expansion ratios of 2:1 for luminance and 4:1 for colour-difference. With this method, the extreme edges of the picture cannot be displayed, although in the receiver the 4:3 window may be moved within the 16:9 picture under the control of a pan signal.

Alternative methods could be used to allow the display of a wider picture on 4:3 aspect ratio receivers; for example, a 'letter box' format with scan adjustment.

SYNCHRONISATION AND CLAMPING

Line, frame and sound/data synchronisation are all provided digitally from part of the data burst at the beginning of each line. An alternative means of frame synchronisation is provided in line 625 by a 64-bit word.

On lines containing a vision signal, a short clamp period is inserted following the sound/data burst.

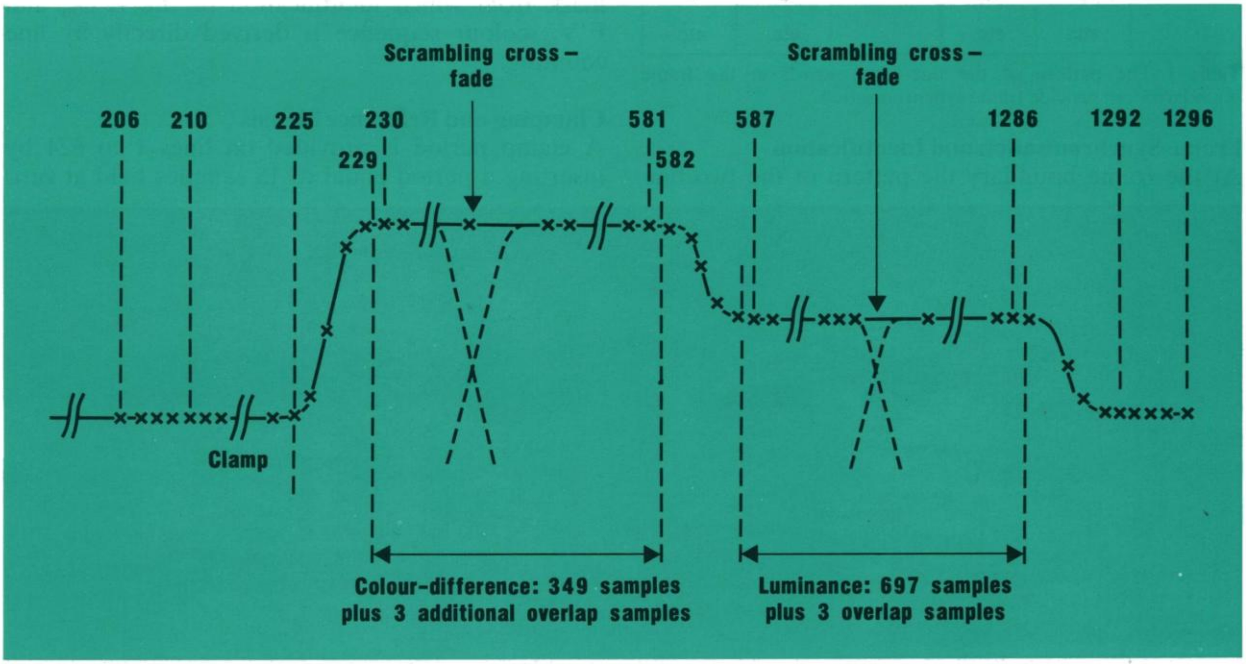


Fig.7. Representation of the scrambled D-MAC video waveform showing cross-fades which conceal the picture edges.

Wider Aspect Ratios

A wide-screen picture with 16:9 aspect ratio may be transmitted rather than the standard 4:3. In this instance the active line-time for the picture remains the same. The presence of the wider aspect ratio picture may be signalled in the wholly data line 625 (by means of a bit 'set' in the 8-bit Multiplex and Video Scrambling Control Group, MVSCG).

A problem could arise when the signals are to be displayed by conventional 4:3 aspect ratio receivers. The picture would have the appearance of being geometrically distorted. This is overcome by selecting a 4:3 window within each colour-difference and luminance component, then in the receiver applying

Line Synchronisation

A 10 microsecond data burst of 206 bits precedes each line (except line 625 where the burst forms the entire 64 microsecond line.) Within the burst the first 6 bits following the demodulator run-in bit form the Line Synchronisation Word.

Frame number	Line number	Sync word	Frame number	Line number	Sync word
	620	W_2		620	W_1
	621	W_1		621	W_2
Even	622	W_2	Odd	622	W_1
	623	W_2		623	W_1
	624	W_1		624	W_2
	625	W_1		625	W_2
Frame Boundary					
	1	W_2		1	W_1
	2	W_1		2	W_2
Odd	3	W_2	Even	3	W_1
	4	W_1		4	W_2
	5	W_2		5	W_1
	etc.	etc.		etc.	etc.

Table 1 The pattern of the line sync. words at the frame boundaries can provide frame synchronisation.

Frame Synchronisation and Identification

At the frame boundary the pattern of the two line sync words can also provide frame synchronisation and identification of the D-MAC two frame sequence. Line 1 of an even frame begins with line sync word W1 and line 1 of an odd frame begins with line sync word W2 as shown in Table 1.

Exceptionally, on line 625, frame identification can also be obtained from a 64-bit Frame Synchronising Word which forms part of the 96-bit Frame Synchronisation Data.

The entire sequence is transmitted in its true form preceding even frames, and in its inverted form preceding odd frames. In addition to its prime purpose, the data can also act as a benchmark from which identification of the $E'U_m$ and $E'V_m$ colour sequence is derived directly by line counting.

Clamping and Reference Signals

A clamp period is provided on lines 1 to 624 by inserting a period equal to 15 samples held at zero

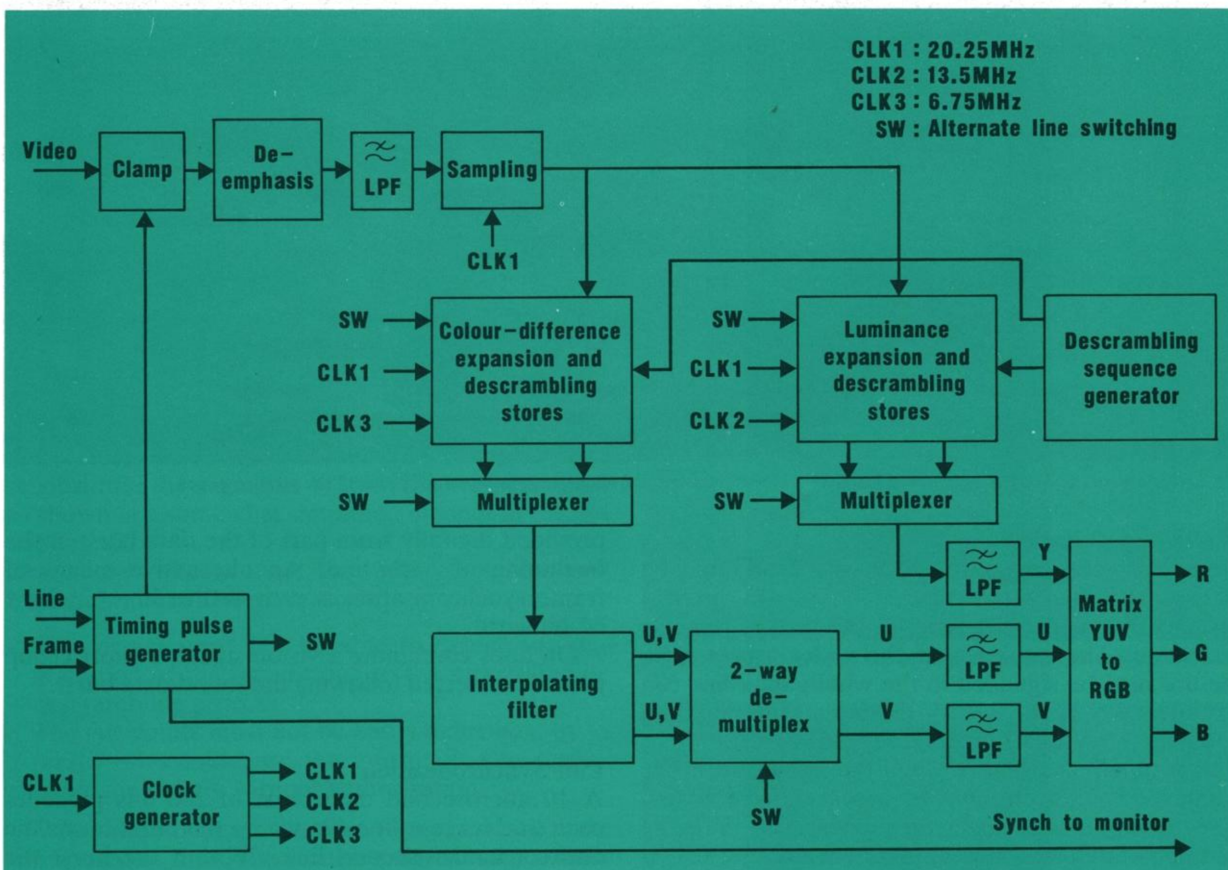


Fig.8. Notional diagram of vision decoder.

level; a 32-bit clamp marker is positioned in the non-standard burst of line 624. To ensure that the marker immediately precedes the clamp period it is positioned such that it is fixed in relationship to the end of the burst, but independent of its duration as shown in Fig. 4.

The vision portion of line 624 is allocated to analogue and digital reference signals.

DECODER VISION PROCESSOR

An appreciation of the vision processing required in the decoder can be gained from Fig. 8.

1. G. Tonge & M. D. Windram. Line-sequential Colour Transmission and Vertical Filtering in MAC.
IBA E & D Report 123/83

Packet Construction and Transmission

Synopsis
The background to possible sound/data coding methods is reviewed prior to a description the D-MAC/packet format. The chapter considers the basic techniques of duobinary coding and how an increase in bit capacity can be achieved. The structure of an individual packet is described including the functions of the packet header together with the occasional need for the 8-bit packet type.

The flexible nature of the multiplex is considered with illustrations to show how an entire frame of two sound/data subframes can support 4100 packets per second giving a net mean bit rate of 2.952 Mbit/s. The alternative uses of the two subframes are also discussed together with the requirements of packet transmission.

Before the action of transmitting the packetised sound/data channel is described, it is useful to consider the background to the choice of data coding and the concept of duobinary coding.

THE CHOICE OF CODING

A major criterion of transmission by satellite is that all the signals should be contained within the 27 MHz r.f. bandwidth allocated to each satellite transmission channel (WARC77). The frequency modulated vision signal, with a baseband bandwidth of approximately 8.5 MHz, satisfies this criterion. Within the 27 MHz channel consideration must also be given to the manner of transmitting the sound/data information. Early decisions in this area resulted in the need to adopt a digital format which, in turn, should relate to the 20.25 MHz vision sampling rate. Alternatives that could be considered include ternary, four-state (quaternary) codes and duobinary codes.

An initial proposal, known as C-MAC, was to employ a form of coding with a 2-4 Phase Shift Key arrangement (symmetrical PSK) which would be placed in an r.f. multiplex with a separately modulated vision format. This method is technically superior to the other proposed formats, providing good error performance in the presence of noise and making full use of the satellite channel bandwidth. Nevertheless, after further consideration a duobinary coding system was found to be a highly acceptable compromise, as it provides an economy of baseband bandwidth and commonality with other proposals made to the European Broadcasting Union.

Within the European Broadcasting Union's family of systems are C-MAC/packet, D-MAC/packet and D2-MAC/packet formats. All these have compatible vision and sound signal coding, but vary in their manner of modulation and it is in this context that the D-MAC/packet format offers the best compromise. In the receiver it requires only a single demodulation process for optimum performance, yet in common with C-MAC/packet it has a sound/data capacity twice that of the D2-MAC/packet format.

DUOBINARY CODING

Historically[1], it was felt that this form of coding could offer a doubling of the bit capacity of a straight-binary system, hence the term 'duo'. Today, however, with other considerations such as noise in the transmission channel, the following comparison is considered to be more accurate:

A duobinary coding technique permits a signalling rate of 20.25 Mbit/s in the 8.5 MHz baseband channel, but a binary coded signal, at a similar rate, would in practice require in the region of 13 to 15 MHz. In addition to this advantage, the duobinary decoder, following the application of a slicing level and with reference to a Viterbi decoding algorithm, is able to sense false '1's' which have been created by noise spikes. However, a simple receiver could still employ the normal decoding method with fixed slicing levels.

The essential feature of the duobinary system is that it is a three-state signalling format as illustrated in Fig. 1.

A series of binary numbers can be translated into a duobinary code by passing the information through the relatively simple circuit of Fig. 2. In the D-MAC coder this comprises an exclusive OR gate and a delay process. The result is a binary bit stream which is changed in the following manner.

With regard to Fig. 1 it can be seen that a binary

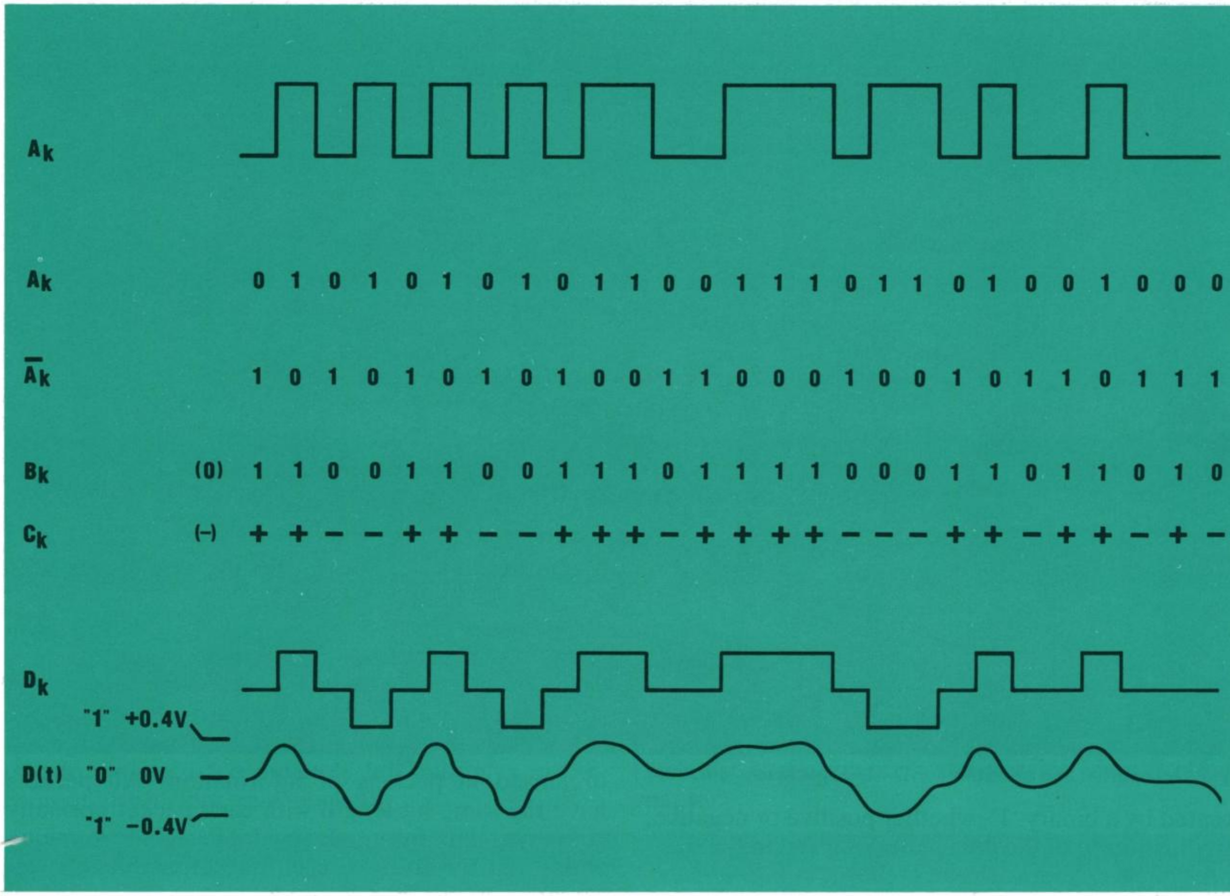


Fig.1. Translation of a binary coded signal to a duobinary waveform.

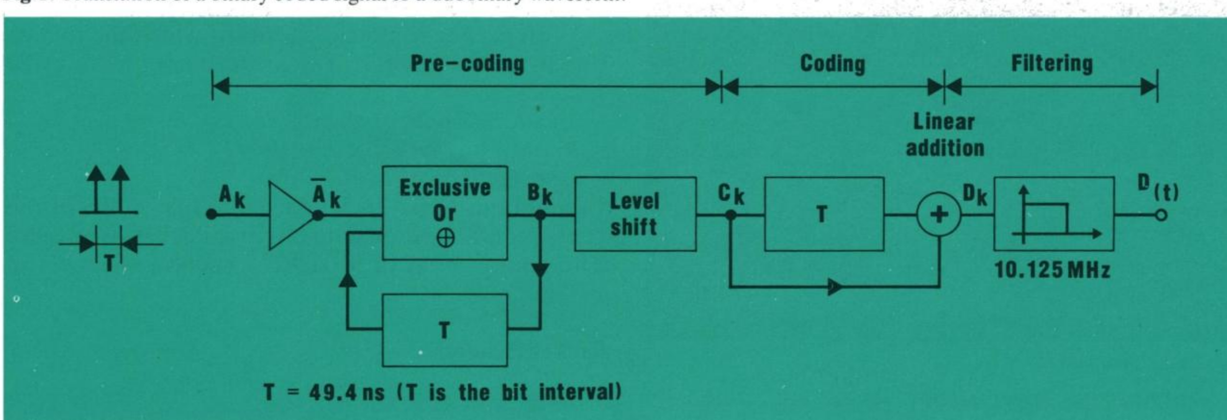


Fig.2. Coding method of a duobinary signal at 20.25 Mbits.

'1' is represented by a positive or negative excursion of the duobinary waveform whereas any '0' is always a zero point. The decision as to whether a binary '1' should be represented by a duobinary '+' or a '–' depends upon the history of the preceding bits. It is also based on the value of the preceding excursion

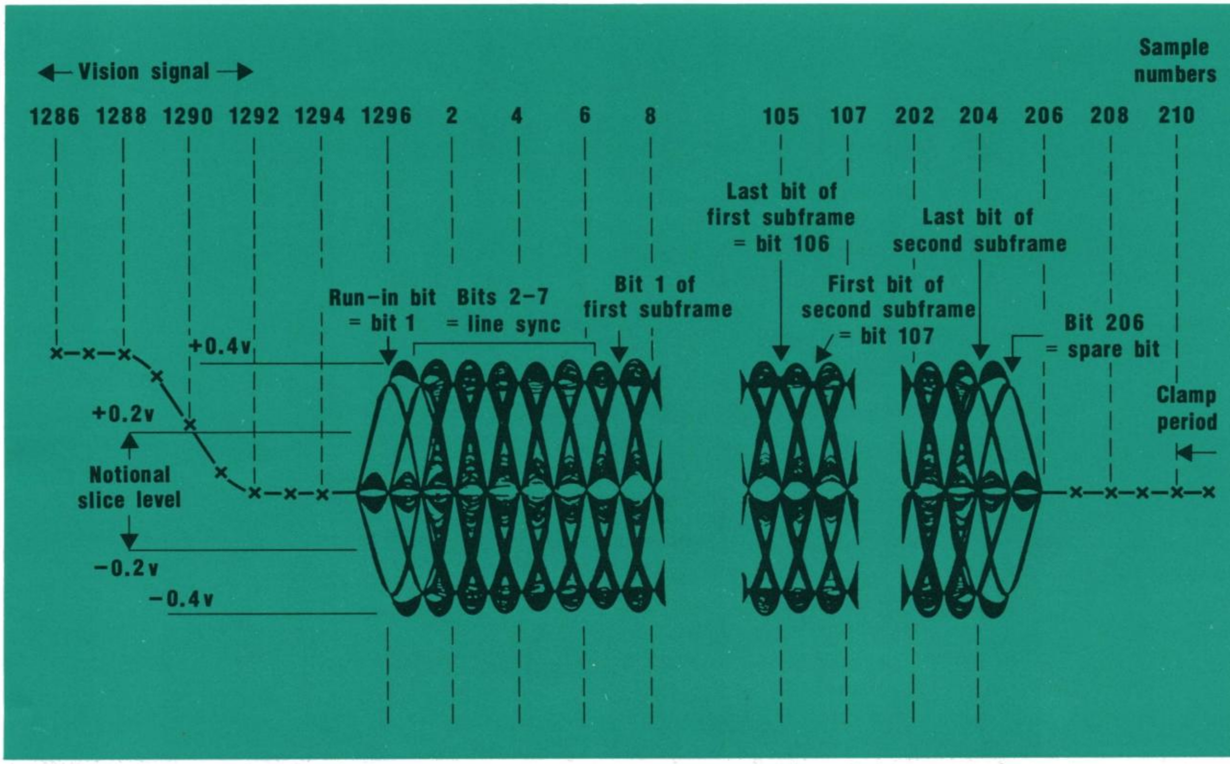


Fig.3. Data period and transition for D-MAC/packet system.

created by a binary '1', whether positive or negative. An illustration of the data period is given in Fig.3.

A typical decoding circuit is shown in Fig. 4 where

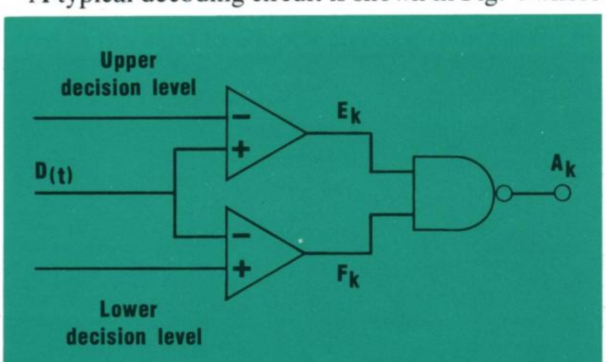


Fig.4. Duobinary decoding.

the upper and lower slicing levels are notionally placed at plus and minus 0.2 volts. The process of translating to a binary coded signal is shown in Fig. 5.

PACKETS

The digital sound and data is carried in a bit stream organised in packets of information. All packets have the same bit length with each packet normally conveying data from only one input signal. A unique address at the front of each packet enables the receiver to accept the desired packets and reject all others.

Each packet contains 751 bits of which the first 23 bits form the header. This in turn comprises three sections:

- an address field of 10 bits
- a continuity index of 2 bits
- a protection suffix of 11 bits

In addition, some packets use the first 8 bits of the remaining 728 useful data bits as a Packet Type byte: The composition of a packet structure is shown in Fig. 6.

Packet Header

ADDRESS FIELD

The 10-bit address field provides for the identification of 1024 possible sound/data services. Packets with address '0' are permanently allocated to the dedicated Service Identification channel. Packets in this dedicated channel convey information giving the user access to the various television sound and data

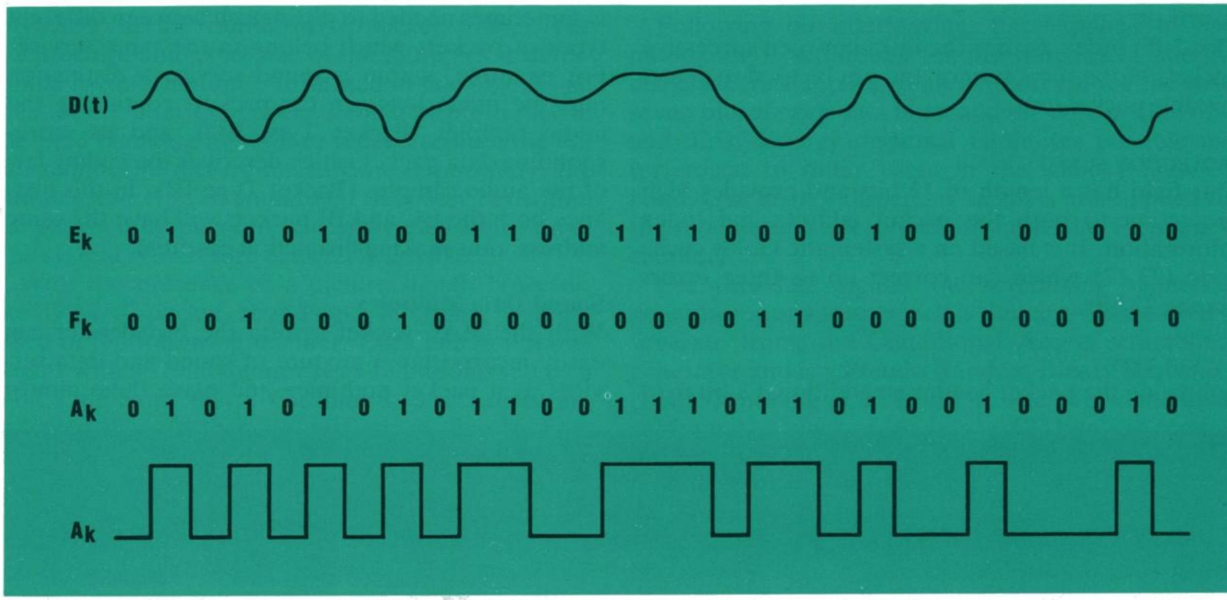


Fig.5. Translation from a duobinary waveform to binary coded signal.

services that are available within the multiplex. (Further details are given in a separate chapter).

Another specially allocated packet address, '1023', is reserved for dummy packets which are used to maintain a completely full multiplex in the event of insufficient 'real' packets.

The receiver can determine all other packet addresses by decoding LISTX in the dedicated Service Identification channel which, as stated, has packet address '0' and therefore is always known. (This list is described in the Service Identification chapter).

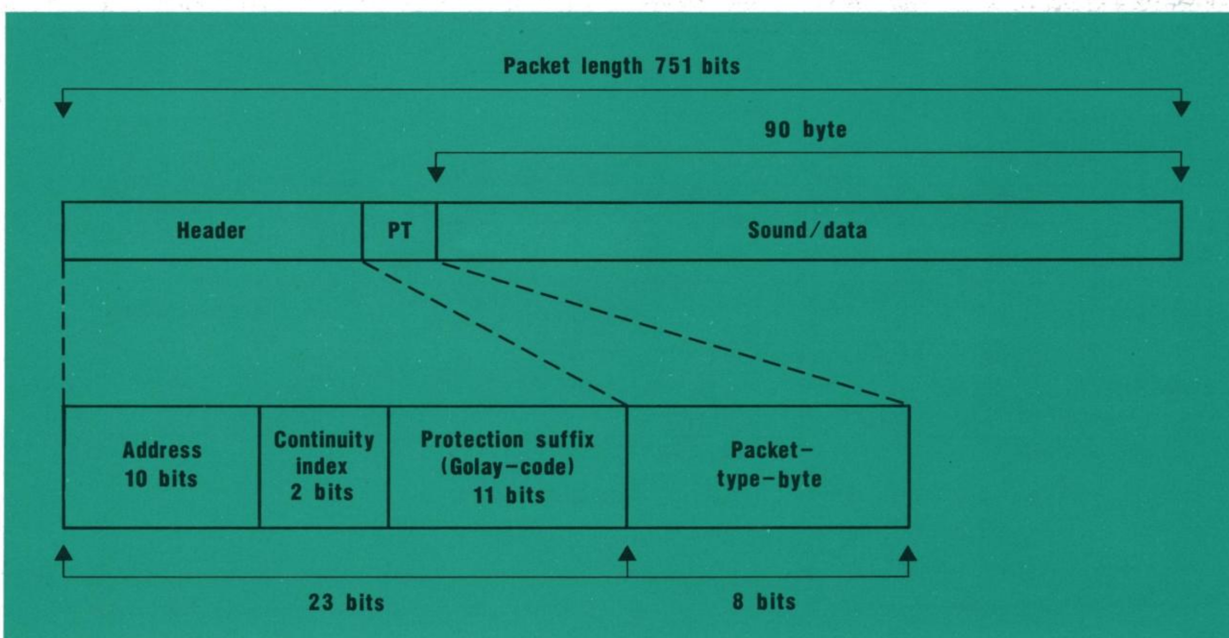


Fig.6. Packet structure.

CONTINUITY INDEX

This 2-bit index assures the links between successive packets of the *same* service and can be used to detect possible packet loss.

PROTECTION SUFFIX

This field has a length of 11 bits and provides high protection to both the packet address and index information. It is based on a systematic Golay cyclic code (23,12) which can correct up to three errors among 23 bits.

PACKET TYPE

Following the packet header an additional 8-bit field is sometimes needed to distinguish between different types of packets which belong to the same service. For example, within a sound service a distinction must be made between the packets containing the audio samples (Packet Type BC), and its corresponding data packet which describes the coding law of the audio samples (Packet Type BI). In this instance both the BC and BI packets will have the same address, thus making distinction essential.

Sound/Data Multiplex

With the MAC/packet system the broadcaster can easily incorporate a mixture of sound and data services as a packet multiplex and place them almost

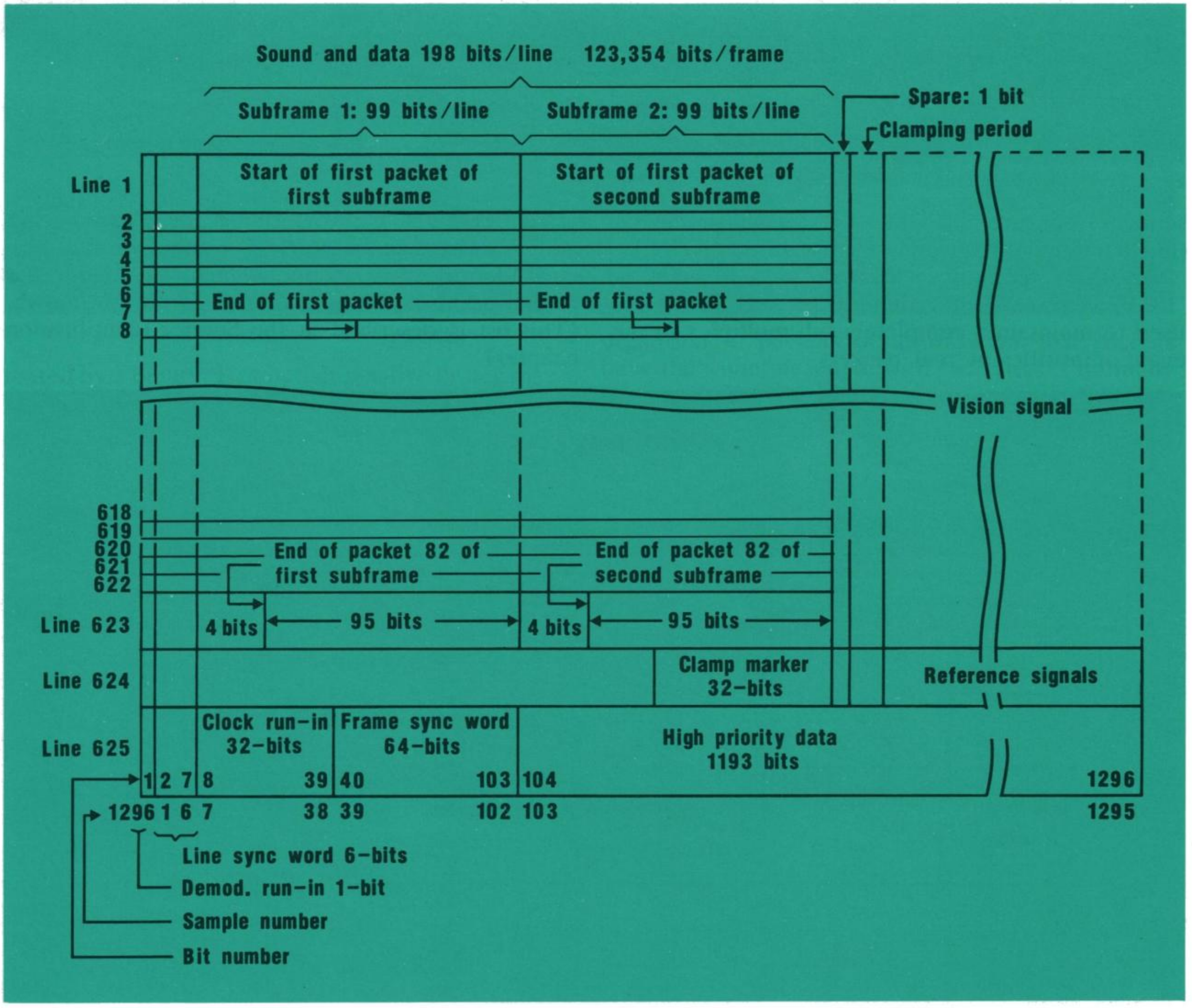


Fig.7. Transmission multiplex structure with packet insertion (not to scale).

anywhere in the entire MAC/packet frame. The architecture and exact size of the multiplex is determined by information transmitted in line 625. As an illustration, a series of packets could be positioned in the field blanking interval or indeed replace the entire number of picture lines. More importantly, the number and type of sound and data services within the packet multiplex can be easily varied at any time. The system is therefore extremely flexible.

With the presence of a picture signal, however, the D-MAC/packet sound/data multiplex is organised in the following manner. Each line in a frame begins with a Demodulator Run-In bit followed by a 6-bit Line Sync Word. On lines 1 to 623 these are followed by a further 198 bits which, line by line, build up over a period of 40 ms into two separate sound/data subframes in the manner shown in Fig. 7. (In fact the complete data burst has a total of 206 bits, but the last bit on each line is unallocated). In this form the sound/data multiplex contains 123,354 bits and since a packet contains 751 bits it follows that each subframe can accommodate 82 packets. Thus over the 40 ms period the total capacity of both subframes is 4100 packets per second giving a net mean bit rate of 2.952 Mbit/s.

Subframe Management

The utilisation of the two subframes can be varied according to the type of digital service and the wishes of the programme provider. In one mode, packets from a particular service may be multiplexed into only one of the two subframes. Alternatively, the packets may be allowed in either of the subframes to obtain the fullest use of the available capacity. The third option is a duplication of services in both subframes.

In the case where related service component packets are allowed in either of the subframes it follows that the entire service can only be recovered by decoding both subframes. In this instance, packets of the same service must never occupy the same relative position in the two subframes so that the decoder will always have to process only one packet at a time.

An additional consideration is that the packets must be in the correct time sequence in the multiplex.

Packet Transmission

Bit interleaving is applied to the 751 bits in each packet in order to minimise the effect of multiple bit errors.

Following bit interleaving, data applied to the modulator is scrambled for the purpose of energy dispersal (this applies to all data except for the first seven bits of each data burst and the data in lines 624 and 625). Energy dispersal minimises possible interference to other users in the same frequency band. (An energy dispersing signal is also applied to the modulator as described in the Radio Frequency chapter).

For the data signals the scrambling sequence generator is of a form which is similar to, but entirely separate from, the Conditional Access scrambling generator and is a Pseudo Random Binary Sequence (PRBS) running at 20.25 MHz. The two sound/data subframes, a block of 198 bits by 623 bits, are scrambled by a modulo 2 addition of the PRBS sequence to the data bits.

At the transmitter the sound/data services are temporarily held in buffer stores and read by a packet switcher to form the transmitted sequence of packets. For example, in the case of a sound service a most important feature of the process is to ensure that there is an unbroken continuity of service. Teletext, subtitles or data services can suffer a more irregular packet transmission rate.

Services which are required to be delivered at an exact time must have a high priority in the packet switching process. An example of such a service would be the Service Management Message (SMM) contained in the Conditional Access system. This service is transmitted in a special relationship to the MAC/packet frame timing.

The following four chapters describe the services conveyed by packets in the sound/data subframes.

Reference

1. A. Lender. The duobinary technique for high-speed data transmission.
IEEE Trans Commun. Electronics (USA) No.66, 214-18 May 1963.

Sound Channels

Synopsis
The capacity of the D-MAC sound/data channel allows up to eight high quality sound channels or sixteen commentary channels, all of which may be accompanied by data with capacity in excess of an existing terrestrial channel. Each high quality channel can have one or two levels of error protection and may be linearly or NICAM coded. This chapter provides a description of these possibilities with a simplified treatment of the NICAM coding option.

The insertion of the various sound channels into the packet structure is described together with overall sound/data requirements.

INTRODUCTION

The highly flexible nature of the sound/data transport mechanism allows a broadcaster to consider a number of sound coding options. When accompanying a vision signal, the D-MAC/packet transmission standard provides, in the sound/data channel, a potential capacity of some 3 Mbit/s. The exact amount of capacity consumed is dependent on the quality of the transmitted sound, the required level of protection given to each individual channel and the over-air signalling requirements.

The digital values of the sampled analogue audio signals are conveyed by packets placed within the two subframes of the sound/data multiplex; further information on this subject is given in the Packet Construction chapter of this review.

A sound channel may be selected from the following options, which result in twelve possible combinations: high-quality stereo, high-quality mono or medium quality mono; linear or NICAM coding; and one of two levels of protection. These are detailed in Table 1.

Type of Sound	
Main programme sound channels (High quality)	Stereo or mono 40Hz — 15kHz bandwidth
	32kHz sampling frequency
Commentary sound channels (Medium quality)	Mono (stereo not envisaged) 40Hz — 7kHz bandwidth
	16kHz sampling frequency
Coding	
Linear coding :	14 bits/sample 2's complement configuration
Near-instantaneous coding (NICAM) :	near-instantaneous companded law with blocks of 32 successive samples
	10 bits/sample 2's complement configuration scale factor on five ranges
Protection Levels	
First level	One parity bit per sample
Second level	5-bit extended Hamming code per sample

Table 1 Choice of sound channels with coding options and levels of protection.

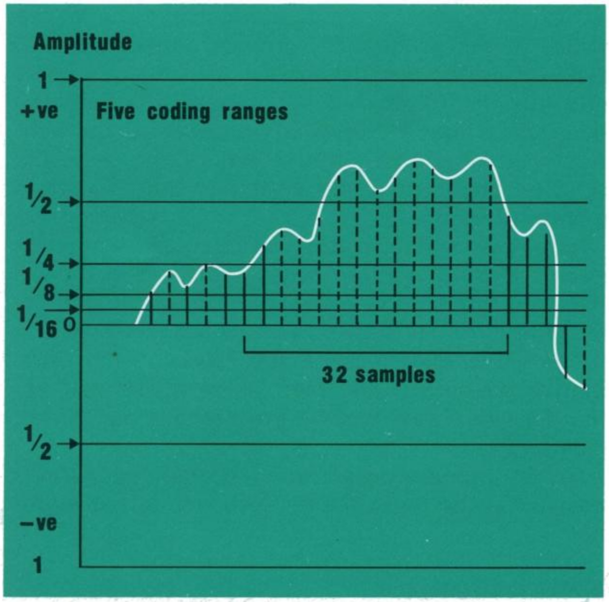


Fig.1. Digitised audio signal of 32 samples divided into a number of binary weighted amplitude ranges. Only the positive excursion is shown, a similar process is applied to the negative excursion (not to scale).

Linear and Companded Sound

Where the transmission of best quality sound is required, a broadcaster may choose to adopt the high quality linear coding option for his main programme sound. This choice, however, even with the application of only first level protection, demands a data rate of 480 kbit/s. The consequence of this is a limitation in the number of such channels and of the data capacity that may be carried in the multiplex.

Companding

Where there is a need for an increase in the number of sound channels, or increased data capacity, the broadcaster may choose to transmit using the Near Instantaneous Companding (NICAM) coding option which reduces the overall bit rate of a sound service. The number of bits per sample is reduced in the coder, transmitted with an appropriate signal (scale factor), and subsequently restored within the receiver. In order to do this, the digitised signal is divided into five binary-weighted amplitude 'coding ranges' as shown in the simplified manner of Fig. 1.

By examining each sound coding block of 32 samples, corresponding to 1 ms of a single full bandwidth audio channel, the smallest amplitude 'coding range' can be selected which will contain the max-

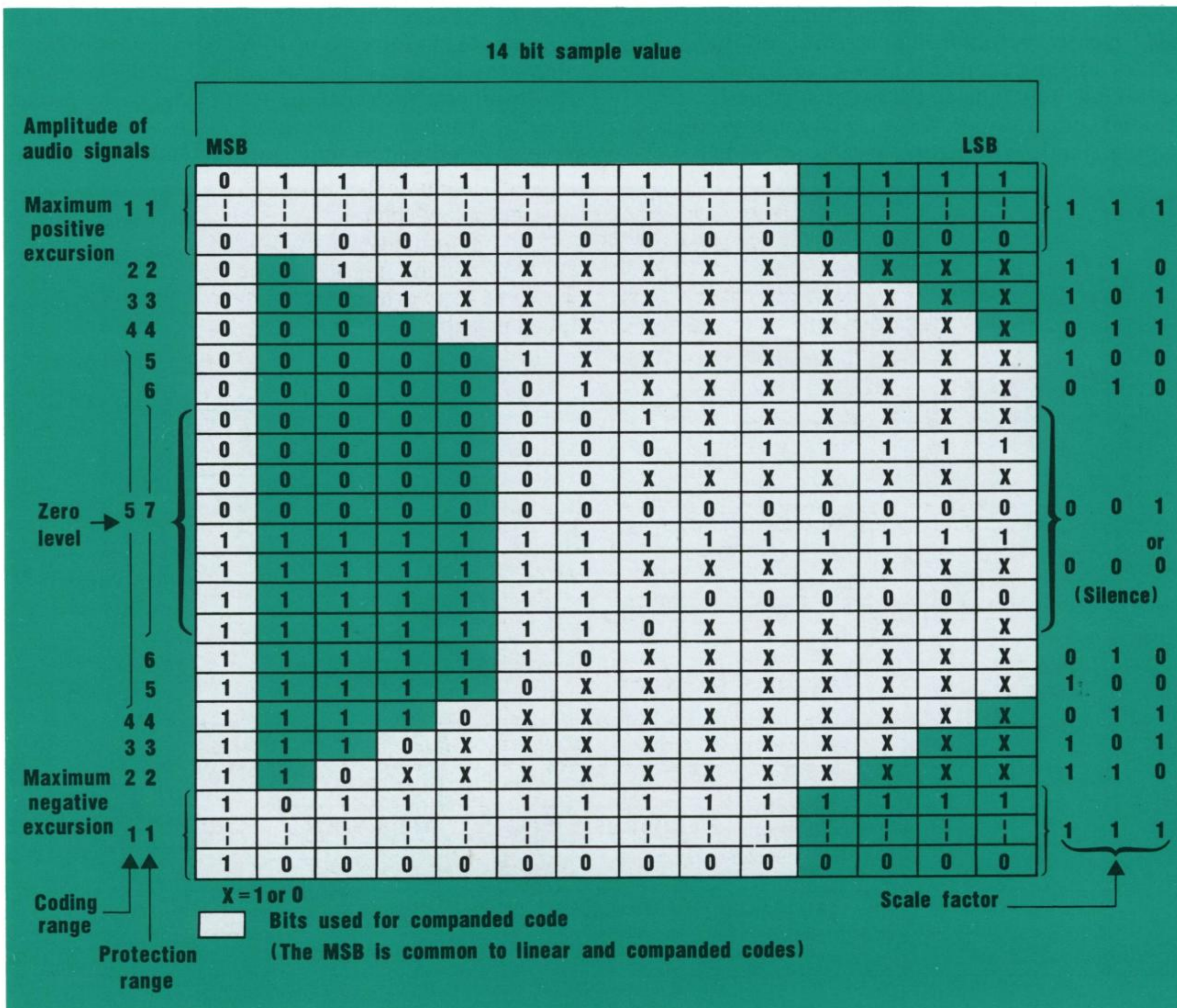


Fig.2. Coding for linear and companded signals (2's complement).

imum excursion of the signal in each block. Four bits are removed from *each* sample in the sound coding block by first deleting leading repetitions of the sign bit and then, if necessary, truncating the least significant bits. The 'coding range' for each block then forms a 3-bit scale factor which is transmitted to the receiving equipment by modifying the error protection parity bits on the audio samples. At the decoder the scale factor is used to restore the missing bits and reform the 14-bit samples of its sound coding block which can then be converted to form part of the analogue waveform.

Linear sound has 14-bit samples and a scale factor is not required for reconstitution of the sample. It is included, however, since during 'quiet' passages the scale factor permits suppression of individual samples at the receiver which have had bit errors thus taking them outside the range specified.

Sound coding samples, with their appropriate coding ranges and scale factors, are illustrated in Fig. 2.

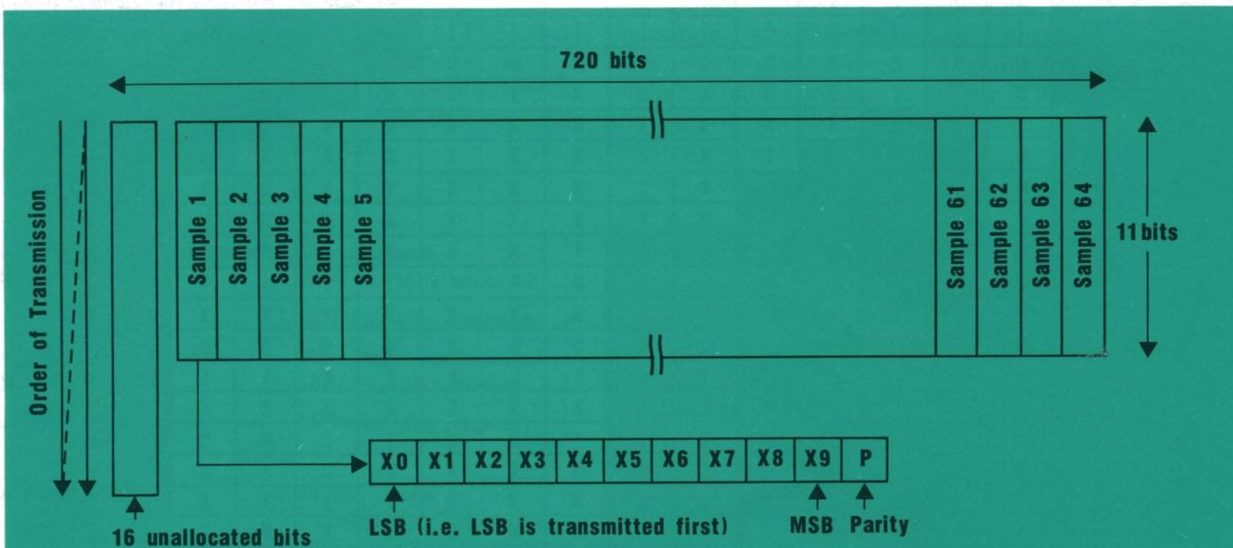
Protection Methods

Two specific protection levels are offered. A basic security level which can be applied to a maximum of eight companded or six linear sound channels when broadcast to a national coverage area. The second option provides a higher level of protection, but has a trade-off in terms of channel capacity reducing the maximum number of sound channels to six companded or four linear.

First Level Protection

COMPANDED SOUND

The first level (basic protection) is achieved by adding a parity bit to each digital audio sample thus allowing the receiver to detect up to one bit error for that sample. In the case of 10-bit companded sound, one parity bit per sample is applied to the six most significant bits such that the sum of the parity group is zero. Thus a stereo signal of 64 samples (32 samples per channel) requiring a total of 704 bits can



	Samples																		
	1	2	3	4	....	29	30	31	32	33	34	35	36	37	....	61	62	63	64
HQ stereophonic companded sound	A1	B1	A2	B2	....	A15	B15	A16	B16	A17	B17	A18	B18	A19	...	A31	B31	A32	B32
	Block n																		
HQ/MQ monophonic companded sound	A1	A2	A3	A4	....	A29	A30	A31	A32	A1	A2	A3	A4	A5	...	A29	A30	A31	A32
	Block n									Block n+1									

Fig.3. Arrangement of 90-byte companded sound coding blocks in relation with first-level protection.

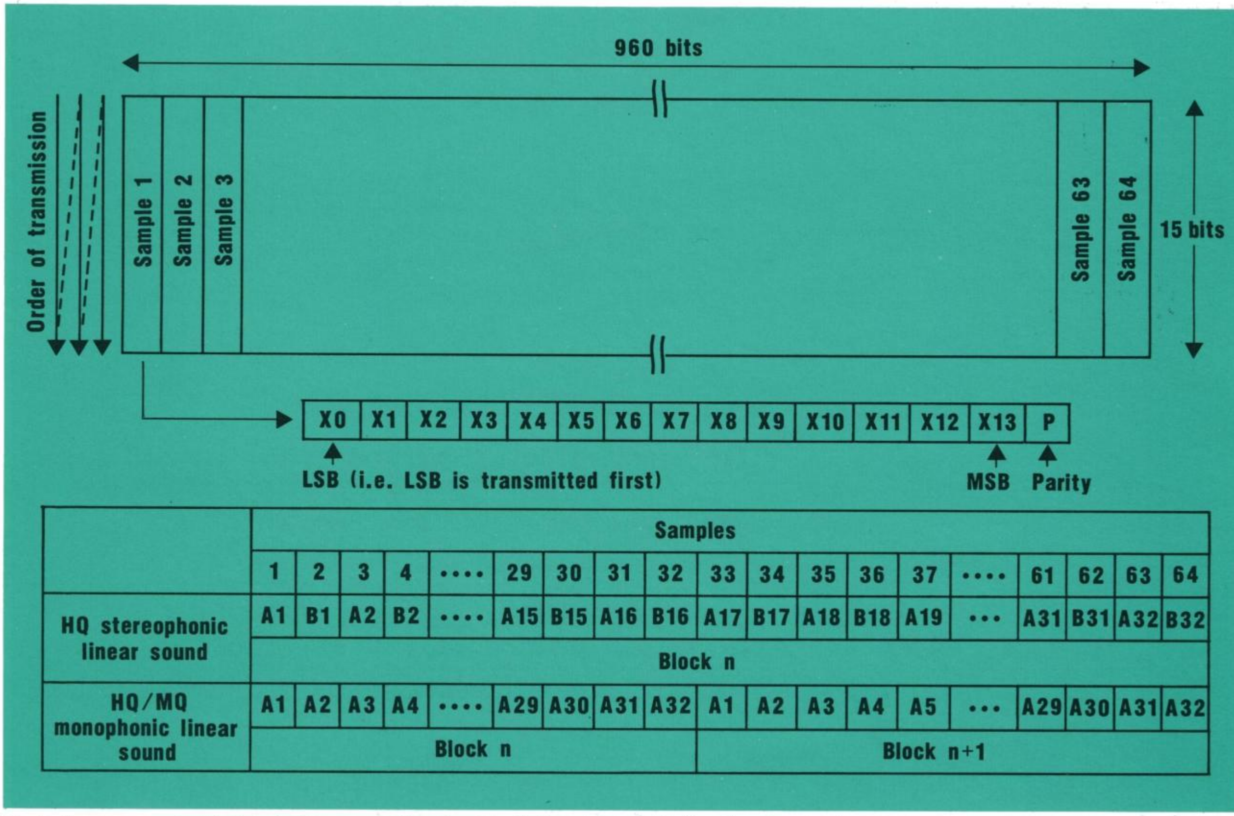


Fig.4. Arrangement of 120-byte linear sound coding blocks in relation with first level protection.

be inserted into a 90-byte coding block. Figure 3. shows this technique and indicates that 16 bits remain unallocated.

LINEAR SOUND

For 14-bit linear sound, the parity bit is applied to the eleven most significant bits in each sample, also to produce an even parity group. In this case with a total of 15 bits per sample a stereo signal now demands 960 bits, that is, exactly 120 bytes as shown in the larger coding block of Fig. 4.

Second Level Protection

COMPANDED SOUND

Second level protection is afforded by applying an extended Hamming (11,6) code to each sample. This permits the receiver to correct one and detect up to two errors in each sample. For the companded audio signals the code is added to the ten bits of each sample to require a 120-byte block as shown in Fig. 5.

LINEAR SOUND

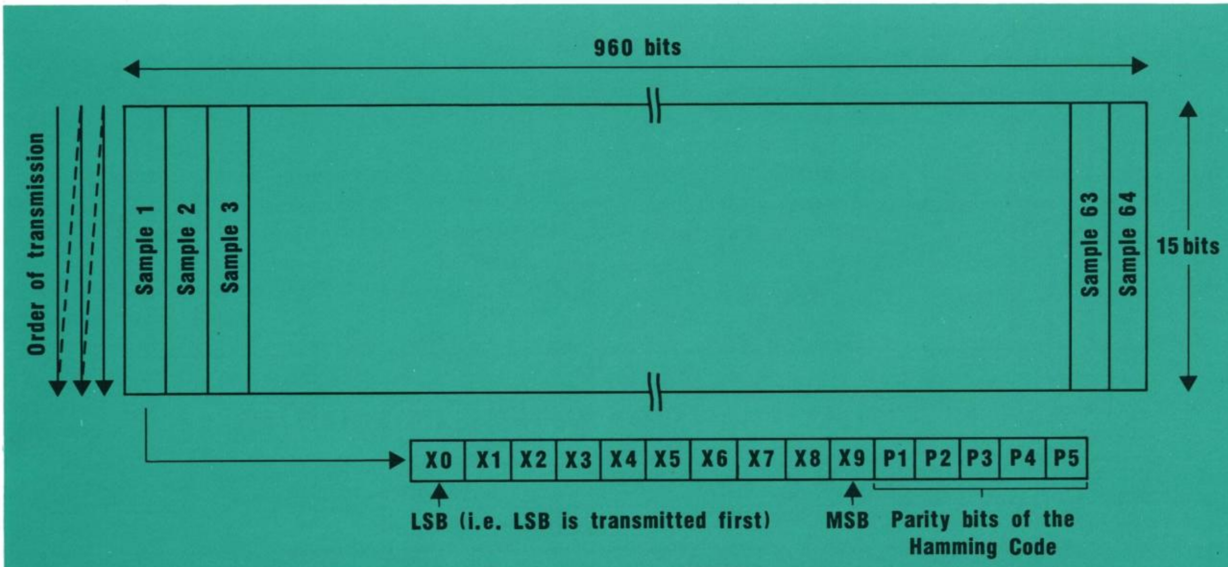
In a similar manner to above, a Hamming (16,11) code is applied to each 14-bit sample of the linear coded signal to construct a 36 sample sound coding block. In this mode, as with the other three sound coding blocks, the scale factor is signalled in the parity bits. However, as this block with its large 'overhead' contains a reduced number of samples, the remaining scale factor bits are signalled in 18 unallocated bits as shown in Fig. 6.

For all four forms of sound coding where stereo signals are to be transmitted, the left and right hand channel samples are interleaved in the fashion illustrated Figs. 3-6.

Insertion into the Packet Structure

The length of the sound coding blocks, as indicated above, depends upon the choice of coding law and protection level:

– companded law and first level protection: 90 bytes
– linear law and first level protection: 120 bytes



	Samples																	
	1	2	3	4	....	29	30	31	32	33	34	35	36	37	....	61	62	63	64
HQ stereophonic companded sound	A1	B1	A2	B2	....	A15	B15	A16	B16	A17	B17	A18	B18	A19	...	A31	B31	A32	B32
	Block n																	
HQ/MQ monophonic companded sound	A1	A2	A3	A4	....	A29	A30	A31	A32	A1	A2	A3	A4	A5	...	A29	A30	A31	A32
	Block n								Block n+1									

Fig.5. Arrangement of 120-byte companded sound coding blocks in relation with second level protection.

- companded law and second level protection: 120 bytes
- linear law and second level protection: 120 bytes

From the above list it can be seen that the two 90-byte coding blocks can be accommodated within a single packet as shown in Fig. 7. However, the 120-byte coding blocks, whose capacity is in excess of a single packet, are inserted in the packet stream by placing three successive coding blocks into four successive packets. This arrangement is shown in Fig. 8, although it should be appreciated that as each packet contains its own address they may be interleaved with other packets in the overall multiplex. Considering the two lengths of sound coding blocks, 120 byte or 90 byte, Table 2 shows the number of audio samples which are placed in a single packet.

Packet Types

The total length of useful data area in a packet structure is 91 bytes (that is 728 bits). For sound transmission the first byte of the data block, known as a Packet Type, is always used to indicate the type of packet (that is, either a sound packet or a sound control packet). Packet Types therefore have two designations: Sound Coding Blocks (BC) and Interpretation Blocks (BI). The Sound Coding Blocks have two possible labels, BC1 or BC2, which when alternated allow precise switching from one coding structure to another (see chapter relating to Interpretation Blocks). The remaining 90 bytes convey either the sound or control information.

Channel Capacity

The data capacity of the sound/data multiplex is considered in detail in the chapter describing the packet structure; but the salient points are that using the two subframes some 4100 packets a second are available, giving a net mean bit rate of 2.952 Mbit/s. This sets the upper limit on the number of channels and data that can be accommodated.

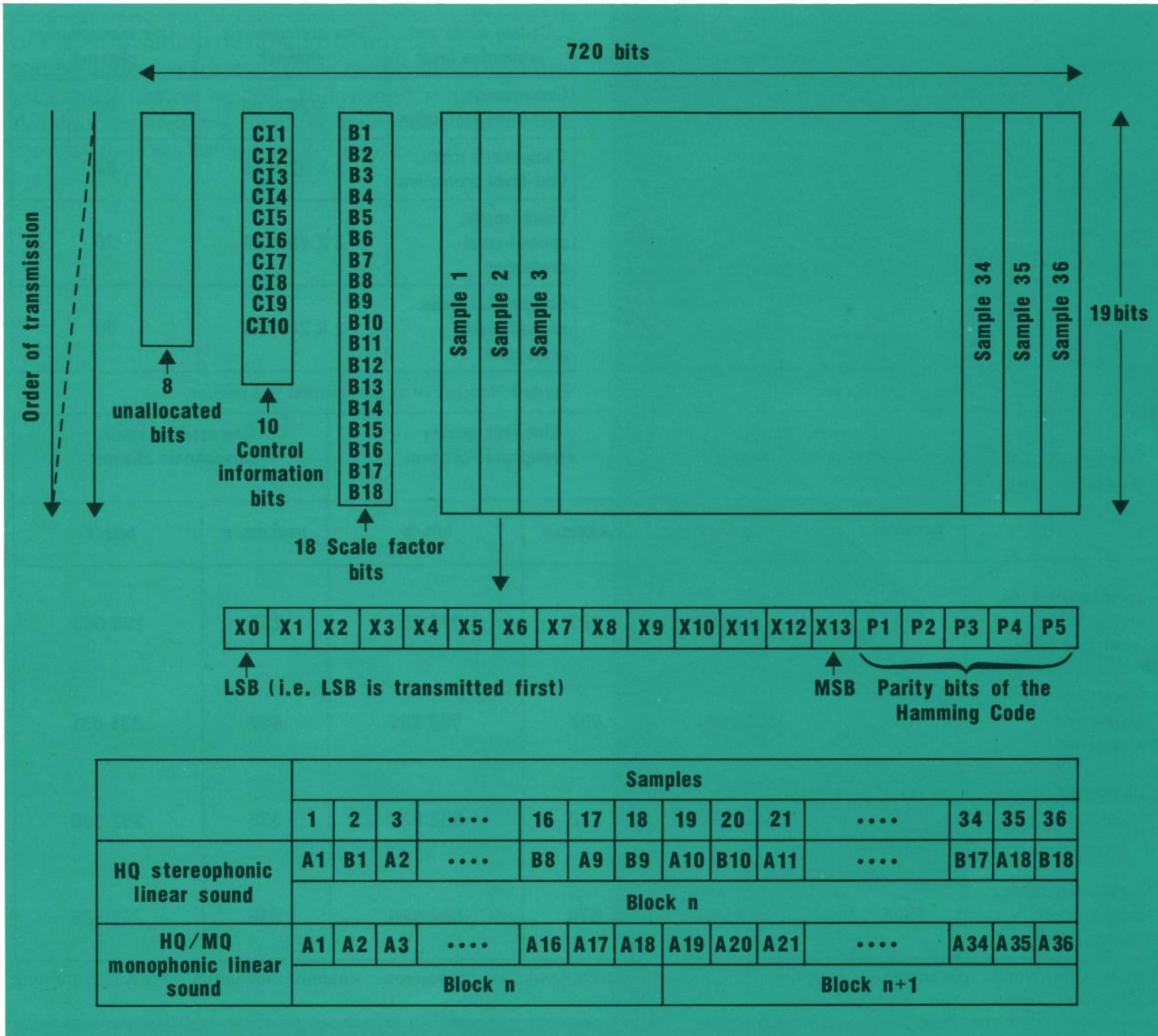


Fig.6. Arrangement of 90 byte linear sound coding blocks in relation with second level protection.

With an audio sampling frequency of 32 kHz, a stereophonic channel and a monophonic channel carry 64,000 and 32,000 samples per second respectively. A monophonic medium quality channel having a sampling frequency of 16 kHz carries 16,000 samples per second. From this information together with the number of audio samples per packet given in Table 2, a notion of the corresponding packet rates can be gained as illustrated in Table 3.

Transmitted along with the sound packets, and with the same address, are additional sound control packets. These contain the Interpretation Blocks and, as mentioned earlier, are identified by Packet Type (BI). Each sound service has its individual Interpretation Block, sent between one and three times per second, which ensures that the receiver has knowledge of the sound channel's coding format. For example, the decoder needs to know whether the audio signal is high or medium quality, if the coding law is linear or NICAM, whether it is scrambled and what level of protection is afforded. These blocks are described in the next chapter.

An appreciation of the sound coding and interpretation blocks together when inserted with other

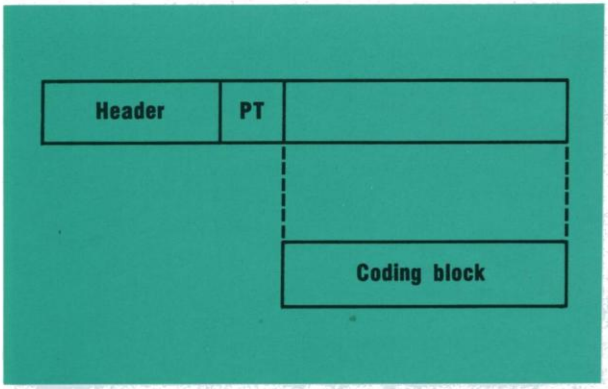


Fig.7. Insertion of a 90 byte coding block.

Coding mode and protection level	One stereophonic channel	One monophonic channel
Linear mode, first-level protection	2 X 24 = 48	48
Companded mode, first-level protection	2 X 32 = 64	64
Linear mode, second-level protection	2 X 18 = 36	36
Companded mode, second-level protection	2 X 24 = 48	48

Table 2 Number of audio samples per packet.

Coding mode and protection level	One high quality stereophonic channel		One high quality monophonic channel		One medium quality monophonic channel	
	packets/s	bits/s	packets/s	bits/s	packets/s	bits/s
Companded mode, first-level protection	1003	753 253	503	377 753	253	190 003
Linear mode, second-level protection	1781	1 337 364	892	669 809	447	336 031
Linear mode, first-level protection	1336	1 003 586	670	502 920	336	252 586
Companded mode, second-level protection	1336	1 003 586	670	502 920	336	252 586

Table 3 Approximate packet rates and bit rates for one stereophonic or monophonic channel, including packets containing Interpretation Blocks.

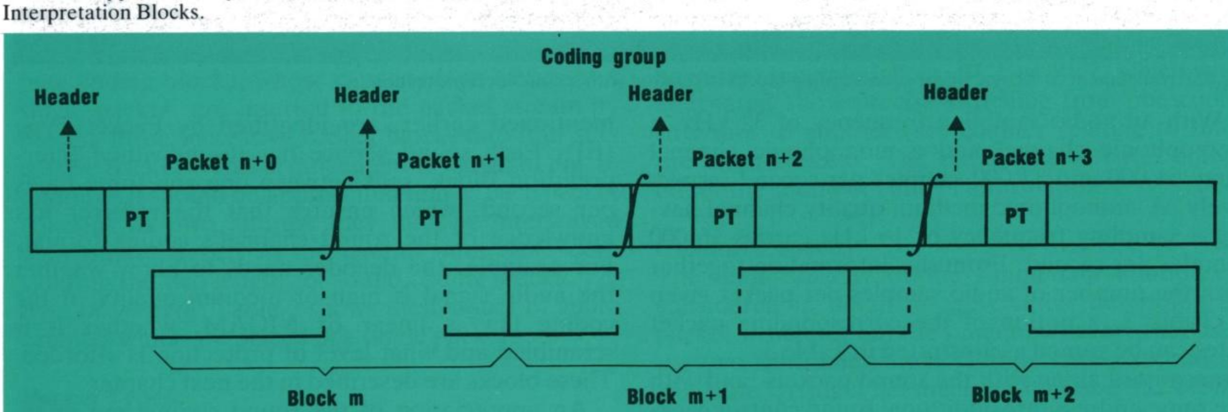


Fig.8. Insertion of a 120 byte coding block.

data into the whole multiplex is given by the charts of Fig. 9. It should be noted that any unused capacity can be allocated for data services. Because the multiplex must always be full, following the insertion of the data services any residual space will be filled with dummy packets having packet address '1023'.

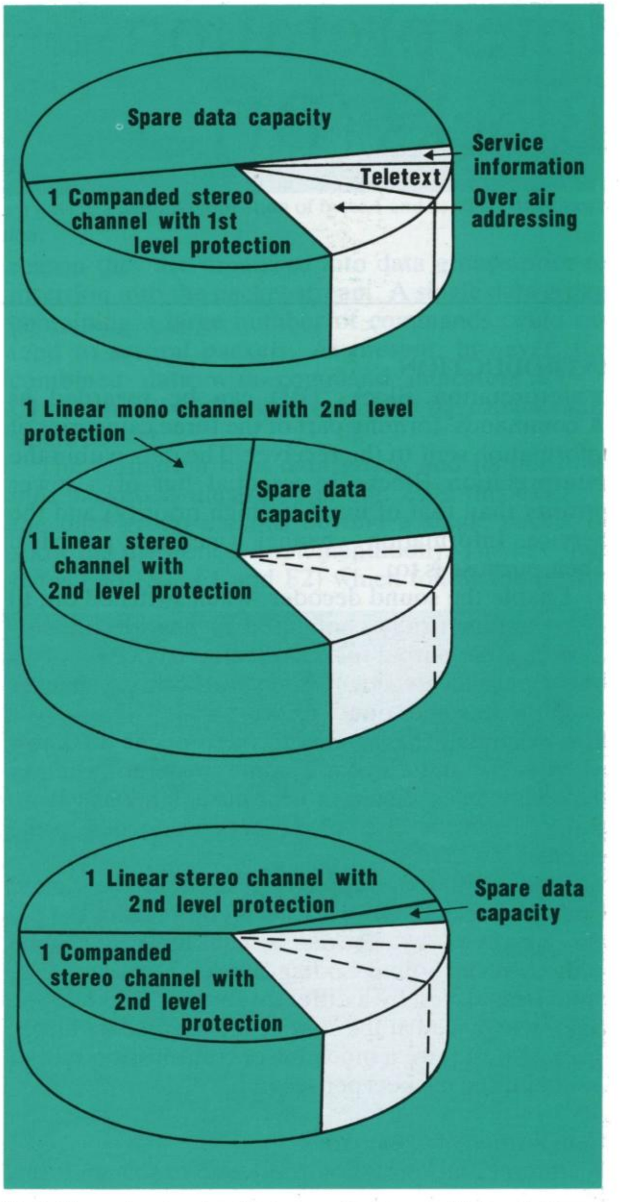


Fig.9. Examples showing the usage of both sound/data sub-frames with a total capacity of 4100 packets/second.

Interpretation Blocks (BI)

Synopsis
The information carried in the Interpretation Blocks enables the sound decoder to automatically provide the correct audio output for the selected service and prepare the decoder for any forthcoming changes in the selected service. This information, in the form of data groups, is conveyed by packets in the sound/data multiplex. The composition of the commands and their organisation into data groups is described together with an example of the Interpretation Block operation.

INTRODUCTION

Interpretation Blocks (BI) can be regarded as 'commands' forming part of the three categories of information sent to the receiver. The data within the Interpretation Blocks is essential but of a lower priority than that of line 625 (high priority) and the Service Information channel (medium priority). Their purpose is to:

a) Enable the sound decoder, when switched on, to be automatically configured to provide the correct audio output for the selected service.
b) Prepare the decoder for any forthcoming changes in the selected sound service.

For example, the sound decoder needs to know whether the audio signal is mono or stereo, whether the coding law is linear or near instantaneously companded, and the level of protection afforded to the channel.

Because they are directly related to the Sound Coding Blocks (BC), the Interpretation Blocks (BI) are conveyed by packets which have the same address as the Sound Coding Blocks. However, they are distinguished by a different form of Packet Type (PT) byte placed at the start of the useful data block. They also require a much lower transmission rate of around three packets per second.

Composition of Commands

Within the Interpretation Blocks up to sixteen forms of command can exist. At present only three are utilised and are identified by a Hamming protected Command Indicator (CI) byte: 'CI = 0', 'CI = 1' and 'CI = 2'. Each Command Indicator is followed by a command Length Indicator byte (LI), also Hamming protected, describing the command length.

The commands indicated by CI=1 and CI=2 contain data needed to control the automatic configuring of the sound coder. The parameter fields of these two commands have the same meaning, but CI=1 always refers to the existing state of the decoder configuration while CI=2 describes a future state.

The composition of these parameter fields is formed from two bytes labelled Byte No.1 and Byte No.2 in the order in which they are transmitted; bit 1 of each byte is transmitted first. Table 1 details the contents of both bytes.

Byte 1:		Byte 2:	
Bit 1	Parity bit check for bytes 1 and 2	Bit 1 Bit 2 Bit 3	Eight audio options 1.Mono.40Hz–15kHz 2.Stereo.40Hz–15kHz 3.Mono.40Hz–7kHz 4–8. Not defined
Bit 2	Flag, if storage of Service configuration is required	Bit 4	Automatic mixing enable
Bit 3	News flash indication	Bit 5	Scrambling status
Bit 4	Sound coding block identification BC1 or BC2	Bit 6	Conditional access status
Bit 5	Sound channels temporal requirement	Bit 7	Coding law status
Bit 6 Bit 7	Selects one of four functions of sound command information	Bit 8	Type of error protection
Bit 8	Flag for sound status		

Table 1 Information conveyed by commands in the Interpretation Block.

The sole purpose of the first Command Indicator (CI=0) is to introduce a list, in ascending order, of those commands in the range CI=1 to CI=15 which are about to be changed (it should be remembered that apart from CI=0 only CI=1 and CI=2 exist at present).

In the case of commands with CI=1 and CI=2, Bytes 1 and 2 are consecutively repeated five times

CI=1 or CI=2	LI=A	Byte 1	Byte 2	Byte 1	Byte 2	Byte 1	Byte 2	Byte 1	Byte 2	Byte 1	Byte 2

Fig.1. Composition of commands with indicators CI = 1 or CI = 2 each having a five-fold repetition of bytes 1 and 2, CI=1 describes a decoder's existing condition, CI=2 describes a decoder's future condition.

as shown in Fig. 1. This technique permits the use of majority-decision logic in the decoder as a means of error protection.

Structure of Data Groups

It is possible that a future use of the Interpretation Blocks may utilise more than the three commands in current use and perhaps require the maximum of all sixteen commands. In such instances some form of mechanism must allow the commands to be spread over more than one block, yet also allow them to be correctly identified and error protected. For this reason they are organised into data groups prior to insertion into the packet stream. A single data group containing a large number of commands could extend to several packets. At present, however, the combined data with command indicators CI=0, CI=1, and CI=2 does not approach the capacity of a single packet.

The formation of a data group and its insertion into packets is illustrated in Fig. 2 (in this example the data group extends beyond a single packet to show the theoretical implementation). The first four bytes (S1, S2, F1 and F2) which follow the packet

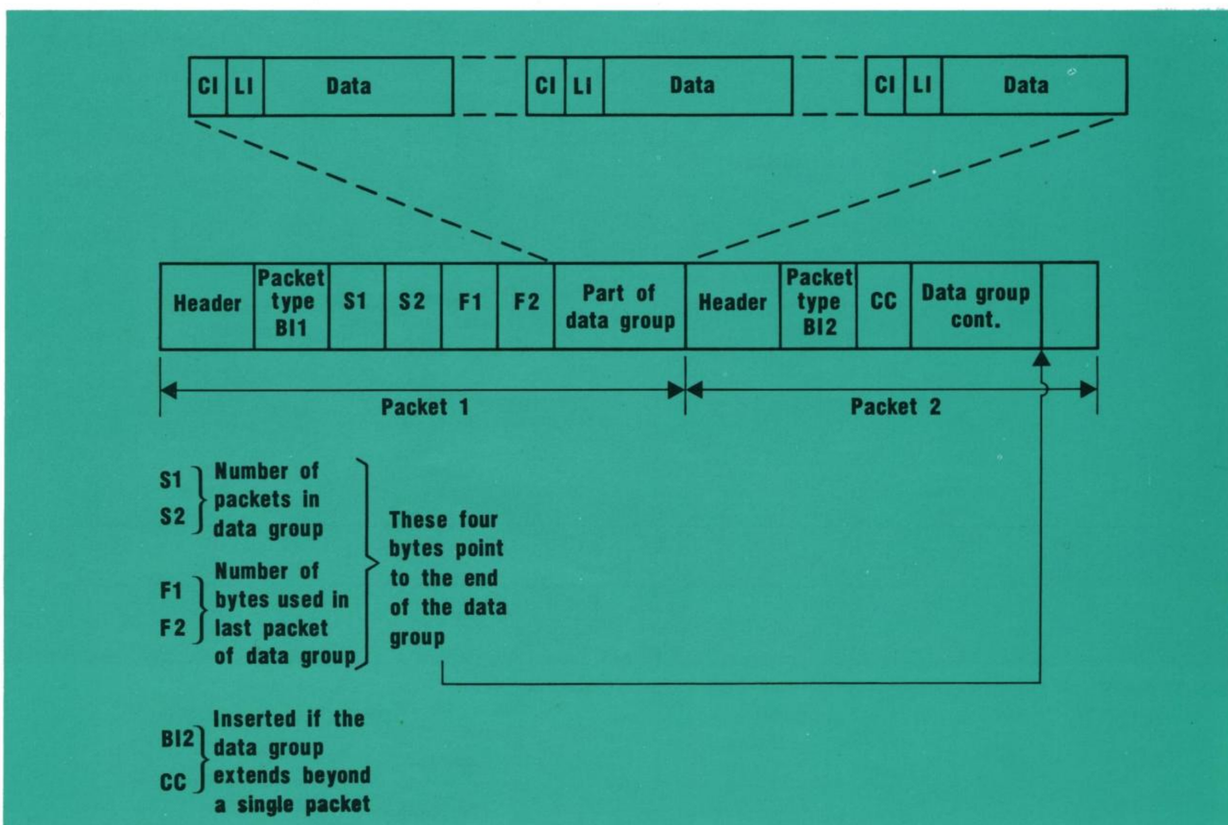


Fig.2. Positioning of the Data Group in the interpretation packets. (This shows the general case where the length of commands causes the Data Group to extend beyond a single packet).

header and the Packet Type (BI) indicate the overall size of a data group together with the number of bytes used in the last packet of that data group. All four bytes are Hamming protected.

Operation of Commands

Under normal steady state conditions the decoder's configuration is described by the parameters contained in Command Indicator CI=1. When a decoder's sound configuration is to be changed, one or more of these parameters must be modified; the procedure is illustrated in Fig. 3. and has the following sequence:

- A data group is formed whose structure will comprise at least three commands.
- A first command with CI=0, whose first parameter byte lists the command numbers which are about to be updated, in this case command number 2.
- A second command with CI=1 describing the existing configuration.
- A third command with CI=2 describing the future configuration.

During the two seconds preceding the transition, the data group containing the new configuration will be signalled at least 3 times per second.

The switching process in the receiver decoder is realised when the Packet Type byte related to the sound coding blocks changes, say from a BC1 to a BC2 or vice versa, for more than a single packet. The new sound configuration takes effect from the third packet after the change. Following this, the parameters within command CI=1 will describe the new configuration and the process reverts to its steady state.

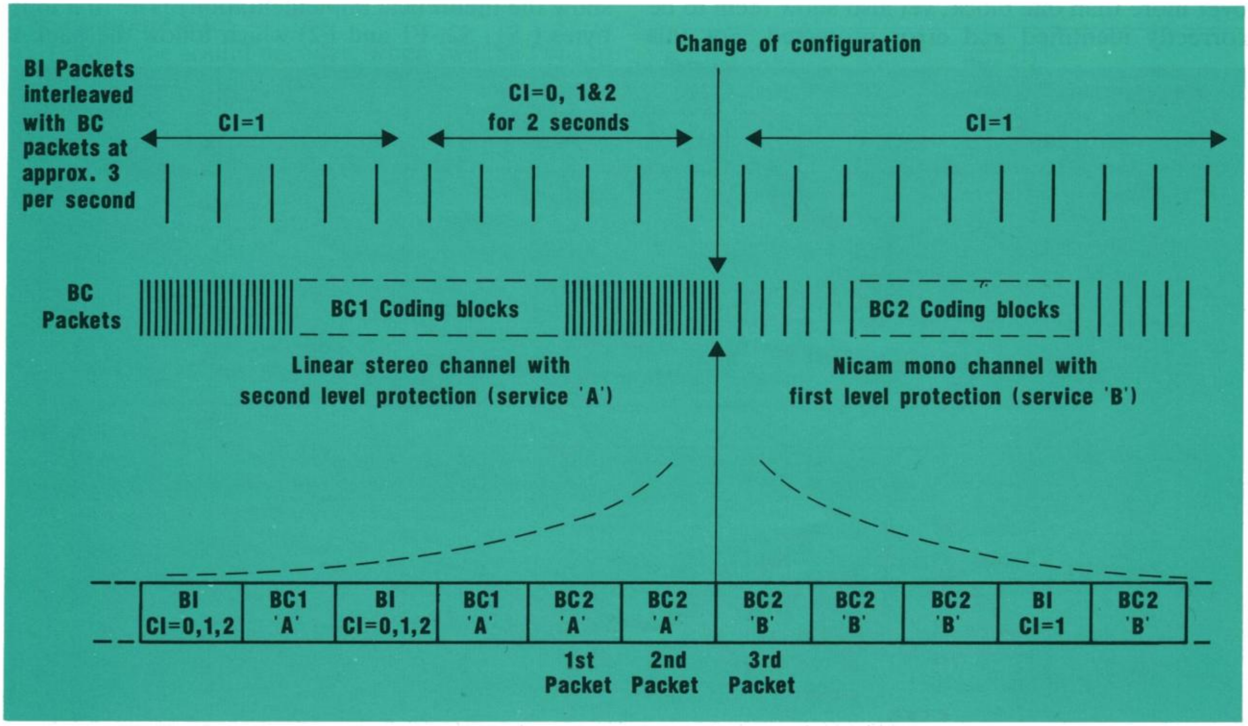


Fig.3. An example of a change in service configuration from service 'A' to service 'B'.

Service Identification Signals Medium and Low Priority

Synopsis
Information about the satellite transmission channel together with details of the vision, sound and data services available are conveyed to the receiver by means of the Service Identification channel. This considerable amount of information is assembled in a tiered manner ranging from the basic parameter of a command, through to the complex format of a Data Group. As with other data in the sound/data multiplex, this information is conveyed by packets. This chapter follows the route of a parameter through the layers of the structure while providing a description of each stage. An example of a typical satellite channel and the capacity required to describe its services is provided at the end of the chapter.

INTRODUCTION

The Service Identification (SI) channel, categorised Medium Level Priority information, gives the user access to the various television, sound and data services that may co-exist in a satellite transmission channel. Once the receiver has gained access to the satellite channel through the High Priority Information in Line 625, the SI Channel will provide a description of the available services together with their components. The system is designed so that most of the information is directly interpretable by receivers. As an example, in conjunction with the High Priority Information (Line 625) and the Lower Priority Information (BI packets), it carries information that allows receivers to be automatically configured to decode the correct service components of a satellite service. An obvious requirement of the SI channel therefore, is that it must be repeated suffi-

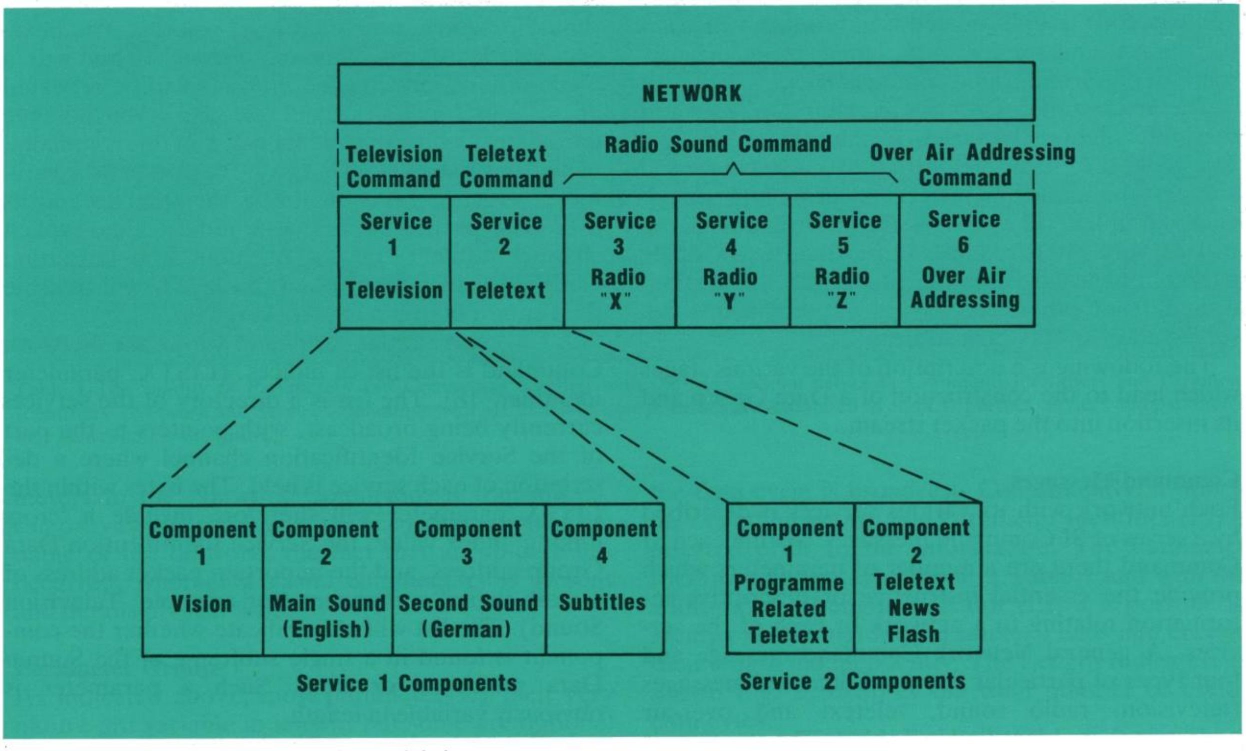


Fig.1. A network comprising six services and their components.

ciently often to allow receivers which have been recently switched on, or those which have changed channels, to rapidly acquire the services.

A complete assembly of *all* the services comprising, television, radio, teletext and over-air addressing is known as the 'Network'. Each service, however, may have several components; for example, a television service obviously contains a vision component, but the vision component may also be accompanied by one or more sound components, in different languages, together with a subtitle component. Within the network other services could coexist, say, in the form of teletext and radio; again each service could have one or more component as shown in Fig. 1.

At a later date it is envisaged that a future generation of 'intelligent' receivers, having collected and edited the service information, could display a 'total menu' of the available services within the network. This could offer the user the opportunity to construct not only a list of programme choice from the network, but also the manner in which the programmes are presented by his receiver. That is, the viewer may choose to receive the main sound in stereo accompanied by a commentary together with a news flash facility. A further example is that the user may opt to decode a sophisticated facility which will allow the main sound service to be faded out in the presence of important sound commentaries.

The amount of data in the SI channel is large and therefore highly structured, forming Data Groups. The groups, in turn, are carried by packets inserted amongst the sound and teletext packets of the sound/data multiplex. The size of the information to be carried very often requires more than just a single packet, but in all cases the SI packets are distinguished from others by having a permanently reserved address of packet address '0'.

The following is a description of the various stages which lead to the construction of a Data Group and its insertion into the packet stream.

Command Messages

Each network with its various services is described by a series of SI Command messages. Within each SI Command there are a number of parameters which provide the essential instructive or descriptive information relating to a network or each of the services. A general Network Command message and four types of particular Service Command messages (television, radio sound, teletext and over-air addressing) are identified in Table 1. The structure is flexible and at a later date further commands may be added for additional control of the system.

Although the Service Identification channel is categorised as being Medium Level priority information, the table shows that the SI Commands themselves are further subdivided having a medium or low priority, M or L.

Parameters

The basic information supplied to the receiver is in a format that varies from parameter to parameter within the command messages. The format is derived from a set of predetermined codes available to the broadcaster. Thus typical parameter field codes would contain a series of bytes representing a description of the network in terms of its services and programme items. This would include the name of the service in 'clear text' for display on the screen, programme type and the language used.

An illustration of a command message with its appropriate parameters can be given by considering the 'Network Command'. Following switch-on, the receiver must quickly identify the correct transmission before configuring itself to decode the accompanying signals. With reference to Table 1, it is seen that to execute this particular command quickly it should be given a medium level priority. The hexadecimal identifying command code is '10 and with a medium level priority the message will be repeated up to four times a second. Of the seven possible parameters which may be included in this command, Network Origin (NWO) can be regarded as a parameter which is fundamental to the receiver and its user. This particular parameter indicates the explicit channel number, orbital position & polarisation; while the coded text, when displayed, will indicate the country of origin in plain language.

A further essential parameter within the Network Command is the list of indices, (LISTX, parameter identifier '18). The list is a directory of the services currently being broadcast, with pointers to the part of the Service Identification channel where a description of each service is held. The bytes within the LISTX parameter will therefore include a 'cross linking' index value, the Service Identification Data Group address, and the important packet address of a main digital component (for example, Television Sound). The list will also indicate whether the component is found in a single subframe of the Sound/Data multiplex or both. Such a parameter is obviously variable in length.

In this particular example the remaining five para-

Parameters identifiers (PI): ·00 ·10 ·14 ·18 ·20 ·40 ·48 ·49 ·60 ·61 ·62 ·78 ·80 ·88 ·90 ·A4 ·A5 ·A8 ·A9 ·AB ·AC ·B0 ·B1 ·BB ·C0

Network and Service commands; Command identifiers (CI); Programme parameters group; Digital component information (DCINF)

Network and Service commands		Priority	Command identifiers (CI)	Update (UPDAT)	Network origin (NWO)	Network name (NWNAME)	List of index value (LISTX)	Local time (TIME)	Service reference (SREF)	Programme item ref. (PREF)	Video programme system (VPS)	Direct commentary (DCOM)	Commentary description (COMD)	Local time description (TIMD)	Access management related messages (ACMM)	Access checking related messages (ACCM)	Analogue TV picture	Digital TV picture (VCONF)	Digital TV original sound	Digital TV additional sound	Digital radio sound	Digital radio additional sound	Digital news flash sound	Digital sound for teletext	Non-cyclic teletext	Cyclic teletext	Digital subtitles	Telesoftware
Network		M	·10	◆	◆	◆	◆					◆	◆	◆														
Network		L	·11	◆				◆				◆																
Services	TV	M	·90	◆					◆	◆	◆		◆		◆	◆	◆	◆	◆	◆			◆				◆	
Services	TV	L	·91	◆					◆			◆				◆												
Services	Radio sound	M	·A0	◆					◆	◆			◆		◆	◆	◆				◆	◆	◆					
Services	Radio sound	L	·A1	◆					◆			◆				◆												
Services	Teletext	M	·B0	◆					◆	◆			◆		◆	◆	◆							◆	◆	◆		◆
Services	Teletext	L	·B1	◆					◆			◆				◆												
Services	Over-air adressing	M	·C0	◆					◆				◆															
Services	Over-air adressing	L	·C1	◆					◆			◆																

Table 1 The validity of Parameters and Commands together with the hexadecimal values of Command Identifiers and Parameter Identifiers. (Network and Service Commands with Medium and Low priorities are annotated M and L). Further services and parameters may be defined in the future.

meters in the Network Command, if transmitted, can provide the receiver with such information as the name of the network, the fact that an up-date has occurred, whether a sound commentary is present and details of local time.

Parameter Groups

As indicated above, most parameters by their very nature are variable in length, but some are optional and others do not need to be transmitted every time the data cycle is broadcast. Consequently, a principle of coding known as a Parameter Identifier (PI) is inserted prior to the information. The Parameter Identifier codes, are shown in Table 1 and with the addition of a Length Indicator (LI) it is possible to identify the numerous Parameter Fields and encode their length independently. The Length Indicator, as shown Fig. 2, denotes the total number of useful data bytes.

A single command message may relate to a num-

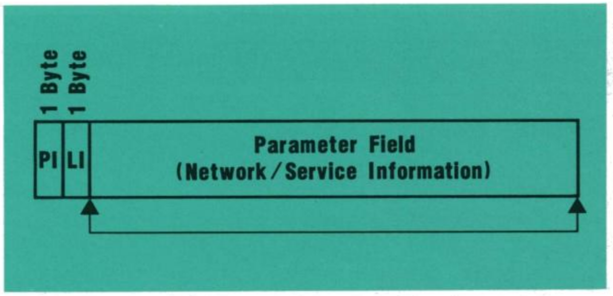


Fig.2. Parameter field containing a variable length of data in bytes which may include coded text, preceded by a Parameter Identifier (PI) and Length Indicator (LI).

ber of services as indicated by the 'Radio Sound' Command in Fig. 1. Consequently, within the command there may be a number of parameters that relate to more than one radio service. It is important that the parameters for each individual service should be kept together. This is achieved by ordering the parameter fields which describe the same service into Parameter Groups (PG). Each group is identified by its own Parameter Group Identifier (PGI), and Length Indicator (LI) as illustrated in Fig. 3.

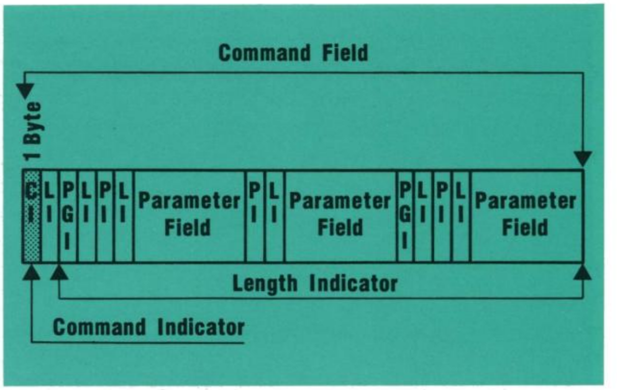


Fig.4. A Radio Sound Command Field showing two Parameter Fields related to the same service and therefore placed within the same Parameter Group. Following is a separate Parameter Field belonging to another service and hence allocated a different Parameter Group Identifier.

Packets and Data Groups

A succession of packets carrying a sequence of Command Fields must be supplemented in order to provide some form of delimitation for the messages together with an error correction capability. This is achieved by creating Data Groups where each group may contain a single Command Field or a sequence of Command Fields according to the size of data as illustrated in Fig.6.

Up to 16 Data Groups each with packet address '0' can co-exist independently and are individually distinguished by a Data Group type number. Data Group type number '0' is reserved to carry parameters that relate to the whole network or where it may give access to parameters that may be carried in other Data Groups. All other Data Group type numbers are allocated by the broadcaster.

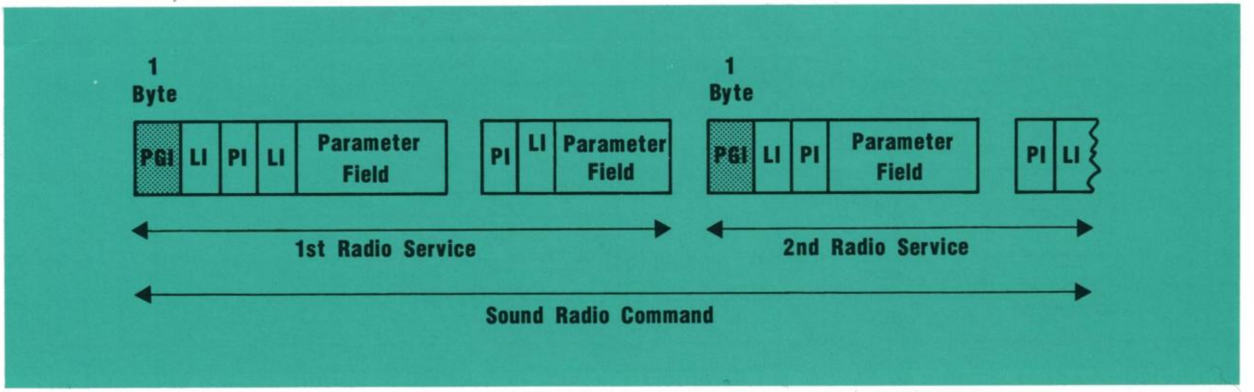


Fig.3. Parameter groups describing two radio services within a single command message.

Command Fields and Sequences

Having constructed and identified the command messages, each containing parameters, a Command Field is formed as shown in Fig. 4.

From these a sequence of commands can be constructed in a similar manner to that shown in Fig. 5.

Command sequences, which are the essential structure of the Service Identification channel, are conveyed by packets within the sound data/burst and for ease of identification have the reserved packet address '0'.

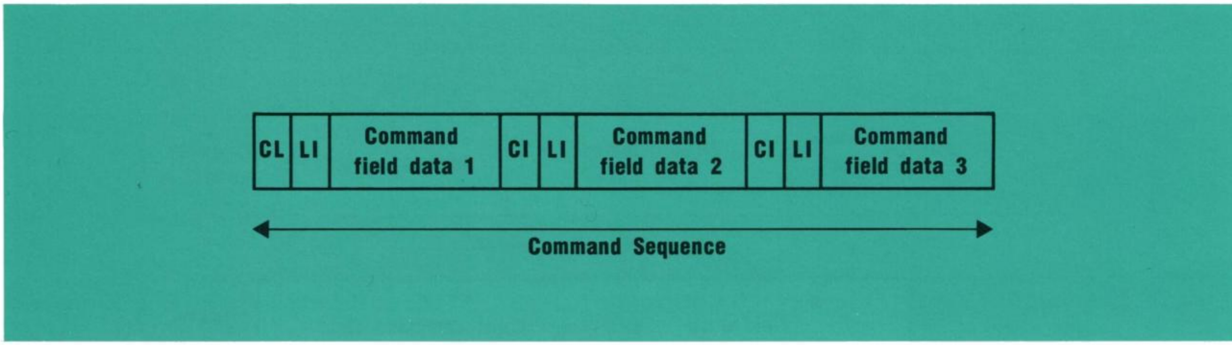


Fig.5. Initial formation of commands into a Command Field sequence.

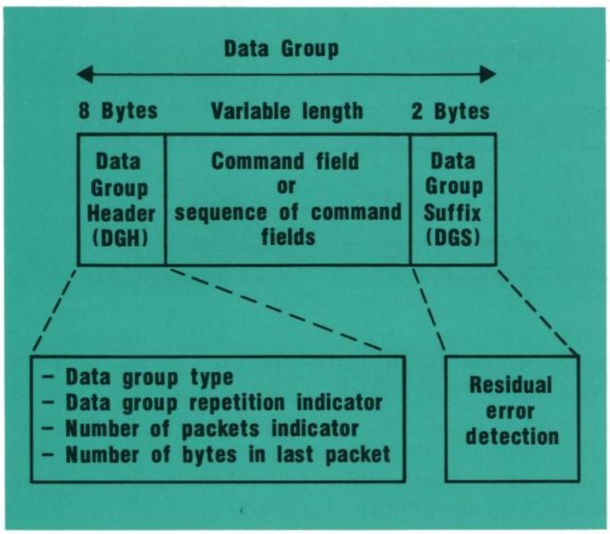


Fig.6. Insertion of a sequence of Command Fields into a Data Group.

Insertion into Packets

Packets which convey the Data Groups of the Service Identification channel, illustrated in Fig.7, have the same structure as other packets in the sound/data multiplex; that is, the useful data portion is preceded by a 23-bit packet header. In this instance the header will always contain the reserved Packet Address '0'.

Error detection at the packet layer is performed by the use of a continuity index in the packet header to detect the loss of packets due to transmission errors, and a packet suffix (PS) to detect errors in the packet.

The first packet conveying any Data Group is designated the synchronising packet which is identified by the special Packet Type (F8). All other packets that comprise the Data Group are allocated Packet Type (C7).

Figure 8 shows the construction of a Data Group from the basic Parameter Field through to the final insertion into a stream of packets.

Channel Capacity and Access Time

An indication of the capacity required to convey the Service Identification channel within a subframe of the sound/data multiplex is given by considering a command message containing medium and low priority information as follows:-

Multiplex 01 (first subframe)

Medium priority command message:

data group 0	(network)	1 packet
data group 2	(TV)	1 packet
data group 4	(radio)	1 packet
data group 6	(teletext)	1 packet
medium priority command message total:		4 packets

Low priority command message:

data group 9	(network)	1 packet
low priority command message total:		1 packet

The medium priority information can be assumed to be repeated four times a second and low priority information once per second, therefore the required mean capacity of the example becomes:

MPX 01 (4 x 4) + 1 = 17 packets/second

The second subframe (MPX 02) could contain similar command messages, say, 14 packets of information per second. Thus the total number

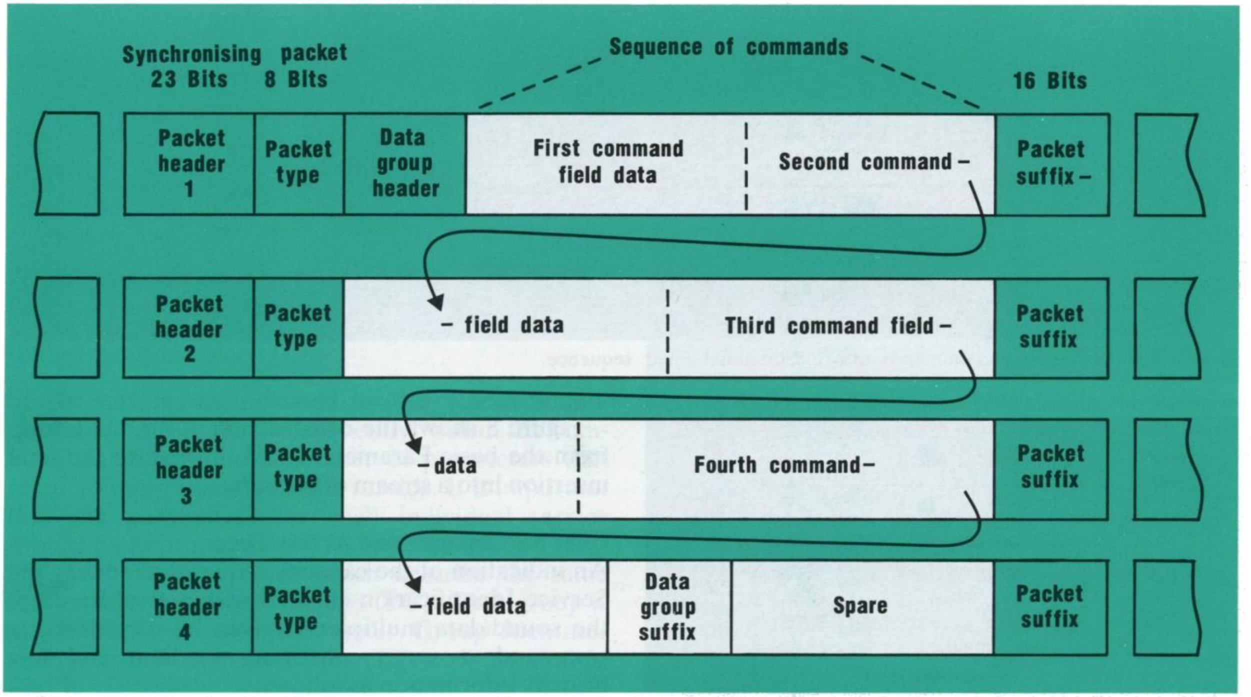


Fig.7. A series of Service Identification packets conveying a complete Data Group in the sound/data bit stream.

of Service Identification packets becomes 31 packets/second within the whole sound/data multiplex. To place this in perspective, the complete sound/data multiplex allows some 4100 packets/second.

The speed at which the Service Identification commands may be acquired by the receiver depends upon the Bit Error Ratio (BER) and the type of Data Group. High BERs could require the receiver to process up to five repetitions which would indicate an access time of around two seconds under poor reception conditions, but substantially less than one second under normal conditions.

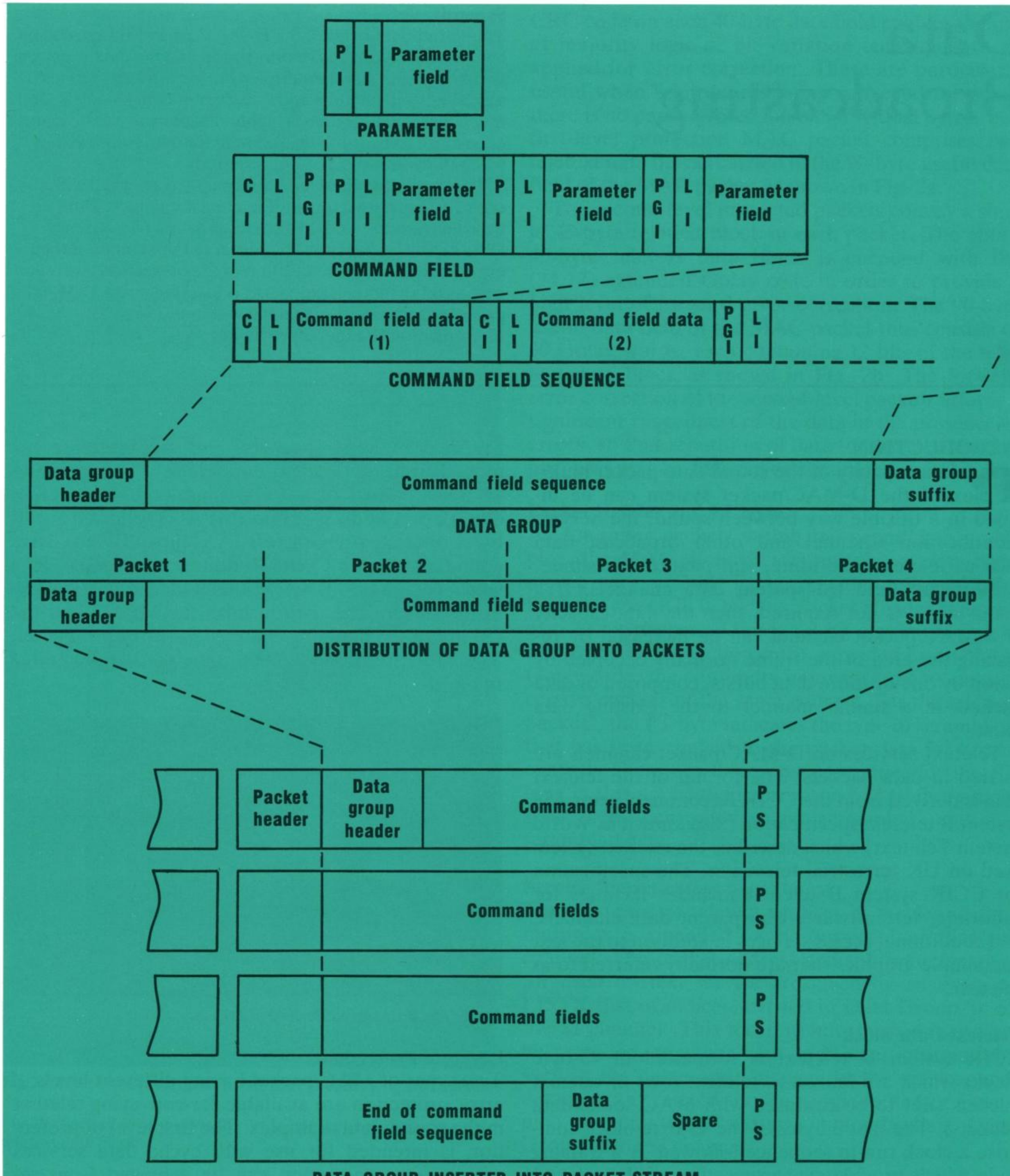


Fig.8. The construction of a Service Identification Data Group from the basic component through to its insertion in a stream of sound/data packets.

Data Broadcasting

Synopsis
The sound/data burst of the D-MAC/packet system can be configured to convey not only sound and the data required to manage the overall multiplex but also to carry teletext, subtitling, still pictures and transparent data channels, and much more. The system is highly flexible and can be made even more so by using the full field for data transmission when the vision signal is not required.

The content of this chapter is mainly concerned with the construction of two forms of teletext data block. Both are derived from the CCIR Recommendation 653 system B (World System Teletext) but differ in the manner of their error protection. A description of the protection levels and how a complete teletext packet is inserted into a D-MAC packet together with signalling within the Service Identification channel is provided.

INTRODUCTION

The data capacity of the sound/data packet multiplex of the D-MAC/packet system can be divided in a flexible way between sound, the Service Identification Channel and other broadcast data such as teletext, subtitling, still pictures, facsimile, telesoftware, and transparent data channels. If a vision signal is not required, then the data capacity of a MAC/packet channel can be increased by replacing the area of the frame normally occupied by vision by one or more data bursts, composed of data packets in a similar manner to the existing data burst.

Teletext services on D-MAC/packet channels are carried in data packets. The format of the teletext data is derived from the CCIR Recommendation 653 system B teletext specification (also known as World System Teletext) which describes the teletext system used on UK terrestrial television. The specification for CCIR system B teletext includes facilities for subtitling, telesoftware, transparent data channels, and conditional access services in addition to the text and simple graphics that are normally referred to as teletext.

Teletext Data Block

CCIR system B teletext is formatted in 45-byte blocks which are known as teletext rows or teletext packets (not to be confused with MAC sound/data packets). The first 3 bytes of the 45-byte block comprise a clock run-in sequence followed by a framing code which provide synchronisation of the teletext data when carried in the vertical blanking interval. These 3 bytes are not required for transmission of the data in MAC packets, and are therefore removed from the teletext data block. The remaining 42 bytes consist of a 16-bit magazine and teletext data packet address group (MPAG) followed by 40 bytes of teletext characters. A Control Byte (CB) is added and then a Cyclic Redundancy Check (CRC) which covers the 40 bytes of teletext characters within the block. The data in both the MPAG and the CB are (8,4) Hamming encoded for error protection. The structure of the teletext data block is illustrated in Fig. 1.

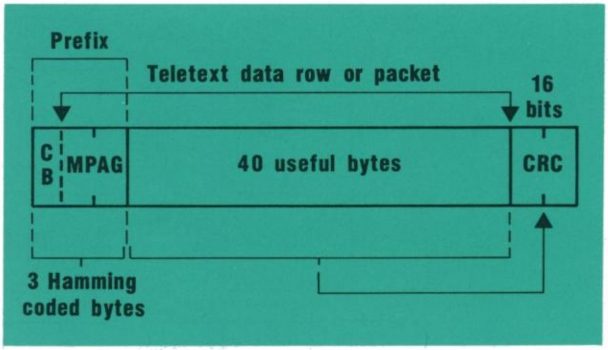


Fig.1. Teletext data block of 45 bytes.

Levels of Protection

Two types of MAC/packet having different levels of error protection are available for conveying teletext in the sound/data multiplex. The first level of protection is intended for use with cyclic data services where error correction can be achieved from repeated acquisitions of the data. The repetitive information in the Control Byte together with the

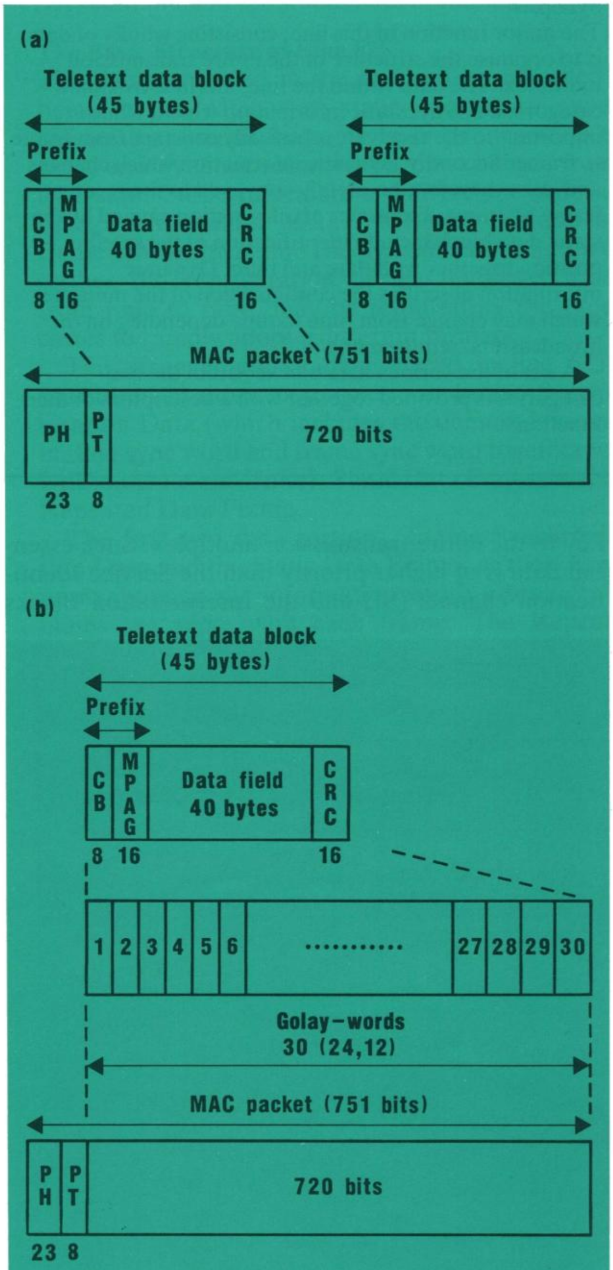


Fig.2a. First level protection MAC packet conveying two teletext data blocks (teletext packets).

Fig.2b. Second-level protection MAC packet conveying a single teletext data block.

CRC code on each 40-byte data field enables the use of majority logic or bit variation techniques to be applied for error correction. These are particularly useful when 8-bit data is being conveyed for which there is no parity protection on individual bytes. The first-level protection MAC packet comprises two teletext data blocks carried in the 90-byte useful data field of the MAC packet as shown in Fig. 2a.

The second-level protected packets convey a single 45-byte teletext block in each packet. The entire 45-byte teletext data block is encoded with the (24,12) extended Golay code in order to provide a high level of forward error correction. The 90-byte useful data field of the MAC packet thus consists of 30 Golay words, each containing 12 bits of the teletext data block as shown in Fig. 2b. The forward error correction of the second-level packets affords a significant ruggedness of the data in the presence of errors, so that repetition of data for error correction purposes may be reduced or eliminated.

Packet Header and PT Byte

As for other packets within the sound/data multiplex, the MAC packets carrying teletext data commence with a 23-bit Packet Header followed by a Packet Type (PT) byte. The Packet Header comprises a 10-bit packet address, a 2-bit continuity index, and an 11-bit protection suffix provided by a (23,12) Golay code; the continuity index increments on successive packets of the same address. For the teletext packets, the PT byte indicates the type of scrambling that has been applied to the packet data for access control. There are three possible values for the PT byte in this application corresponding to free-access scrambled, controlled-access scrambled, and unscrambled data.

Service Identification for Teletext

The existence of teletext services in the sound/data multiplex is signalled in the Service Identification channel. The packet address and subframe location for each teletext service are given by an entry in LISTX (list of indices) carried in Data Group '0' of the SI channel. This locating information may also be provided, together with additional information, such as the name and language of the service, and details of any related service components (for example, for conditional access), in a subsidiary Data Group of the SI channel.

High Priority Information – Line 625

Synopsis
The major function of this line, consisting wholly of data, is to organise the structure of the entire transmission multiplex. The data within the line falls into two main categories. Firstly, static information which, while important to the receiver, is basically constant from frame to frame. Secondly, dynamic information which changes and must therefore be rapidly conveyed to the receiver frame by frame. Examples of information carried by the static data block include identification of the satellite position together with date and time. Dynamic information describes the configuration of the multiplex which may change from time to time depending on the broadcasters' requirements.

A list of the various data blocks within the two categories is provided together with a description of their functions.

INTRODUCTION

Line 625, as presented in Fig.1, contains wholly digital information which can be regarded as the key to the entire transmission multiplex. Such essential data is of higher priority than the Service Identification channel (SI) and the Interpretation Blocks

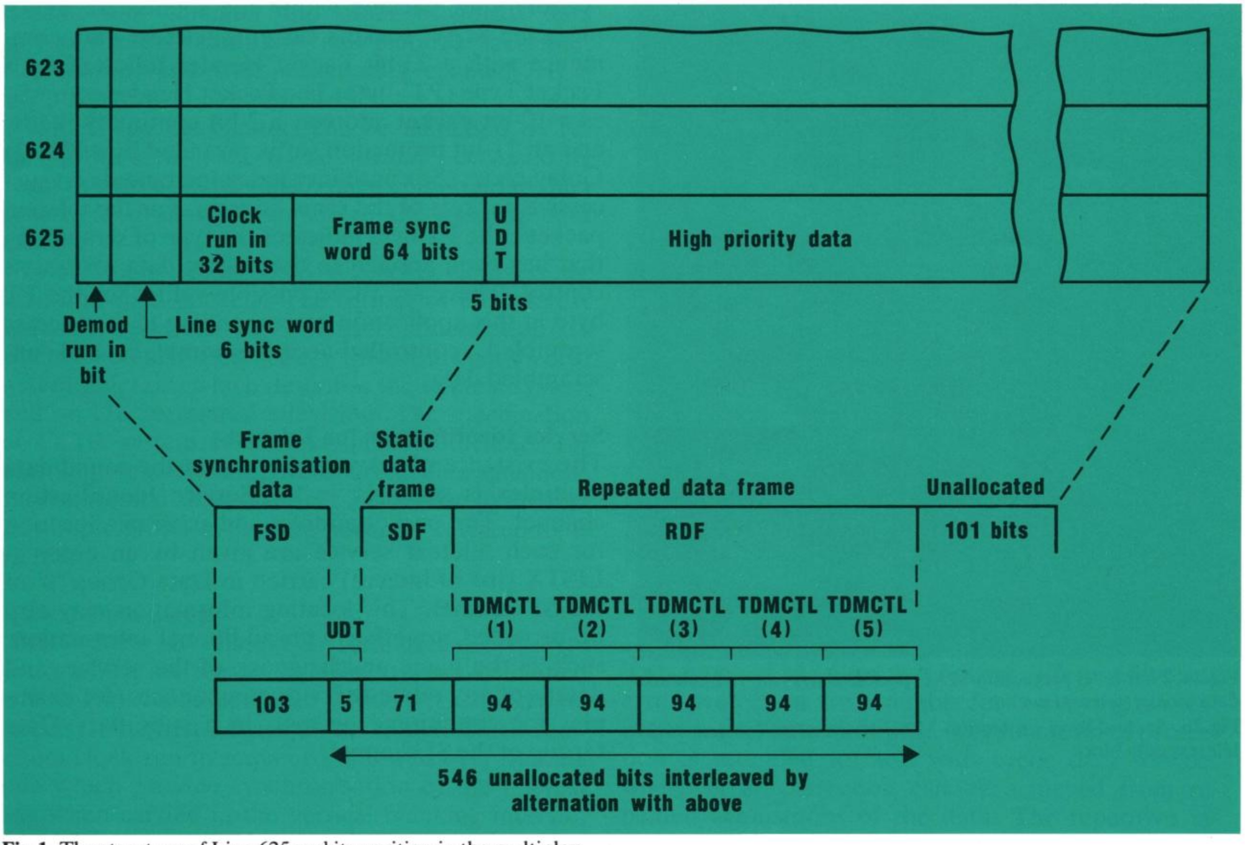


Fig.1. The structure of Line 625 and its position in the multiplex.

(BI) contained in the sound/data multiplex.

The Basic Structure of Line 625

The data contained in the line can be regarded as being in two categories; static information, which essentially remains the same from one frame to the next, and dynamic information that changes frame-by-frame. Static information will include date and time, identification of the satellite's position and frame synchronisation. The dynamic information describes the configuration of the multiplex which, from time to time, may be required by the broadcaster to rapidly adopt a different format.

The arrangement of data placed in line 625 is structured in three major sections; Frame Synchronisation Data (which includes the demodulator run-in, line sync word and frame sync word together with Unified Date and Time), Static Data Frame, and the Repeated Data Frame.

The first two data sections, Frame Synchronisation Data and the Static Data Frame, as their names imply, contain the information which essentially remains the same with each frame. The Repeated Data Frame, however, describes components which can have a changing format and which must be rapidly indicated to the decoder. To aid this, and improve error correction in the receiver, the data is repeated five times per frame.

Both the Static Data Frame and the Repeated Data Frames contain a 14-bit error control group which can be used to detect most error patterns and to correct one or two errors in their respective groups.

A more detailed description of each section of data is given in the Appendix.

APPENDIX

The component parts of line 625

Frame Synchronisation Data (FSD)

Frame Synchronising Data comprises 103 bits which include the demodulator run-in bit, and the 6-bit line synchronisation word (both common to each line of the multiplex), and a 96-bit frame synchronising sequence (see Fig. 2, which also shows the demodulator run-in bit).

DRI 1 bit	Demodulator Run In: Common to all data bursts.
LSW 6 bit	Line Synchronisation Word: Common to each line in a burst, it can provide an alternative for video frame synchronisation by its pattern at each frame boundary.

The entire 96 bits of the frame synchronising sequence, which comprise the CRI and the FSW, is transmitted in its true form preceding even numbered frames and in its inverted form preceding odd numbered frames. Consequently, if required, the two frames of the MAC sequence may be identified from the Frame Synchronising Data.

CRI 32 bit	Clock Run In
FSW 64 Bit	Frame Sync. Word

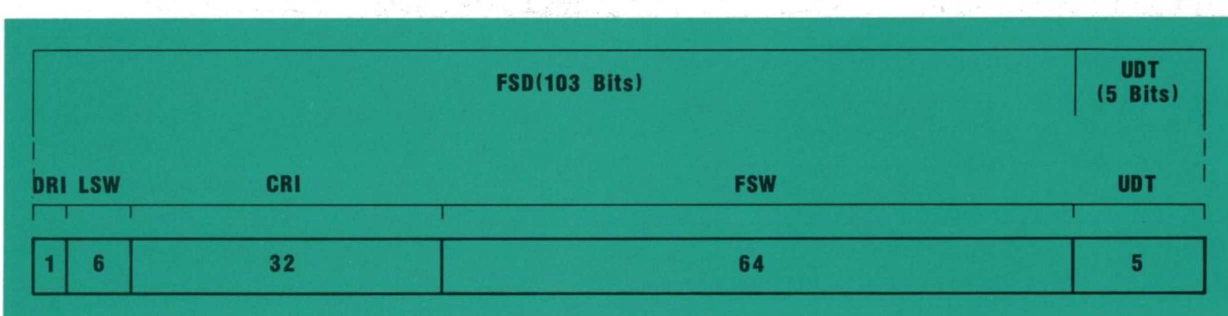


Fig.2. Frame Synchronisation Data (FSD) with Unified Date and Time (UDT).

Unified Date and Time (UDT)

Five bits of data which change from frame to frame over a 25-frame sequence to build up a statement containing the Unified Date and Time. This sequence is modelled on an internationally accepted format of the date and time which has provision for a local time offset. The information is not error protected, but can be reliably determined by the local clock acting as a 'fly wheel'.

Static Data Frame (SDF)

The frame contains four subsections of the format shown in Fig.3; it is used for information about the channel and about the time division multiplex format, together with information to assist the normal operation of the sound and television decoders. In general, all of this information is repeated many times, so majority logic can be used to assist the correct recovery of the data under high error-rate conditions.

CHID 16 Bit	Satellite Channel Identification Code. A unique world-wide code which can indicate the satellite position, channel number and polarisation, country of origin and the responsible administration.
SCR 8 Bit	Services Configuration Reference. This 8-bit feature allows a broadcaster to identify the commonly used configurations of sound services within the channel. Suitable decoders can then be set to allow rapid reception of the main sound of the television service and major components of frequently used radio services. Details that may be stored by a receiver include packet address, scrambling, error protection and audio configuration. The provision of this feature is optional and only applies to the main sound services and not to commentaries or additional languages.
MVSCG 8 bit	Multiplex and Video Scrambling Control Group. This comprises two sub-groups: Time Division Multiplex Configuration (TDMC) and Vision Scrambling and Access Mode (VSAM) which give information on the physical signal organisation within the satellite channel.

TDMC	
bit 1	–indicates if 'normal' time compression is being transmitted.
bit 2	–indicates the presence of a normal sound/data multiplex format.
bit 3	–If set to '1' only the single subframe (01) is recommended to be transferred to the D2 format. When set to '0' either subframe may be transferred.
bit 4	–A '1' indicates that the normal aspect ratio of 4:3 is being transmitted. A '0' signifies the 16:9 aspect ratio.
bit 5	–Unallocated.

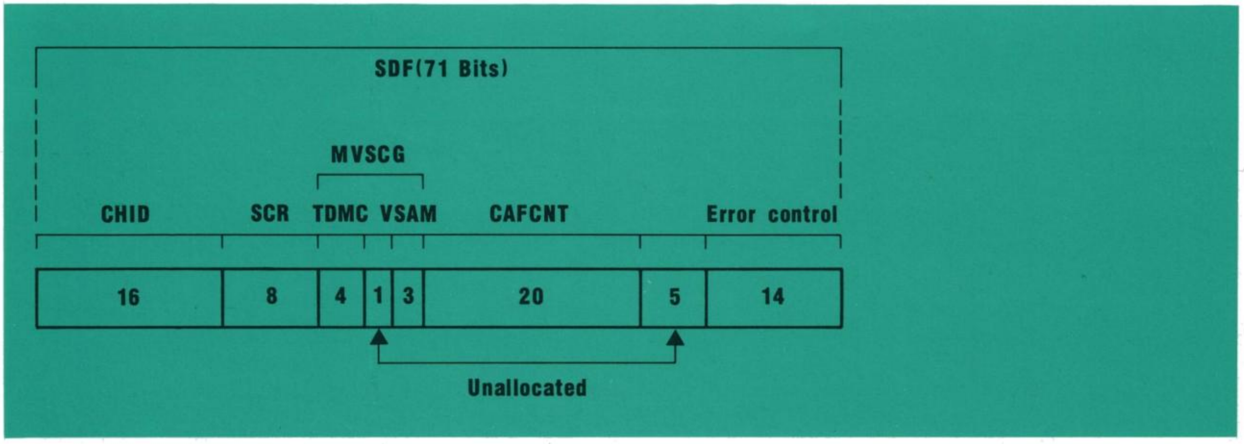


Fig.3. Static Data Frame (SDF).

VSAM 3 bit	Vision Scrambling and Access Mode. These 3 bits are used to indicate whether the programme provider is allowing free or controlled access.
CAFCNT 20 bit	Conditional Access Frame Count. This section contains the 20 most significant bits of the 28-bit frame count, and increments every 256 frames. (The eight least significant bits are carried in the Repeated Data Frame.)

At the end of this sequence 5 bits are set to '1' and are unallocated. The last 14 bits provide error control.

Repeated Data Frame (RDF)

The frame (illustrated in Fig. 4) consists of a fivefold repetition of identical 94-bit Time Division Multiplex Control (TDMCTL) blocks. This important data contains the configuration of the whole of the Time Division Multiplex by defining the size and position of the of *all* its components: vision, data bursts, teletext, and data in the vertical blanking interval and also which lines are allocated to reference signals.

FCNT 8 bit	Frame Count. As its name implies, the code provides a cumulative 8-bit frame count utilising the LSB's of the frame counter. A modulo-128 count is used to synchronise changes in the TDM configuration and to delimit groups of television frames as given below. A modulo-256 count is used in conjunction with the conditional access system and the CAFCNT of the Static Data Frame.
UDF 1 bit	Up-date Flag. Indicates whether the RDF contains new information describing the structure of the corresponding TDM component.
TDMCID 8 bit	TDM Component Identification carries a unique code for every type of Time Division Multiplex component, the size and position of which are defined by the TDMS part of the Repeated Data Frame. The codes are in hexadecimal format: 00 Unallocated 01 The first area within the television frame reserved for sound/data (normally taken to be the first subframe). 02 The second area in the television frame reserved for sound/data (subframe two). 03-0F Other areas within the television frame reserved for sound/data.

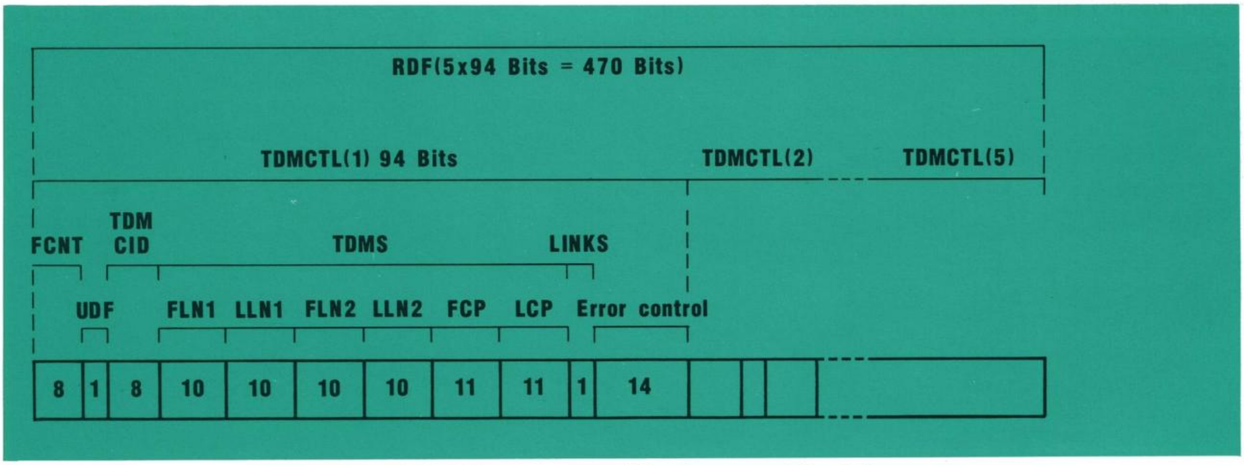


Fig.4. A single Repeated Data Frame (RDF). (This frame is repeated five times across Line 625).

10	Identification of the colour difference components ($E'U_m$ and $E'V_m$).
11	Identification of the luminance component (Y).
12-1E	Reserved for future vision applications.
1F-21	Areas not used by D-MAC.
30-3F	Reserved for test signals in the field blanking interval.

TDMS 62 bit	Time Division Multiplex Structure. Six codes which define the horizontal and vertical boundaries of the subframes in terms of line numbers and clock periods of the TDM component described by TDMCID.
LINKS 1 bit	Linked Structures. A one-bit switch used to link a group of TDMS fields that may be required to fully define one TDM component.

The Conditional Access System

Synopsis
Satellite transmissions may be rendered unintelligible to the unauthorised user by a process commonly known as scrambling. However, scrambling alone is insufficient if a pay-per-view format is desired by the channel operator. The signals themselves may be scrambled, but some device is needed to lock and unlock the scrambling process; this is known as encryption. The chapter describes both these processes and the manner in which the unlocking 'keys' can be passed through separate transmission channels to the receiver. A pay-per- view system is discussed followed by a practical implementation of a shared key over-air addressing system together with a suggestion of how to eliminate the pirating of programmes. The transmission and data capacity requirements are also considered.

INTRODUCTION

The programme provider is able to choose whether to send the source components unscrambled (in 'clear' form and therefore free access) or scrambled. DBS transmissions to the United Kingdom using the D-MAC system will be scrambled.

A scrambling process, however, does not necessarily form a complete conditional access system. The type of conditional access used in the D-MAC system has two parts, a scrambling section and an encryption section. Broadly, the scrambling section is the part which processes the programme signals, while the encryption section processes the key signals to unlock the programme. As such, the scrambling process is implemented essentially in hardware and is therefore fixed. The form of encryption and degree of sophistication, however, can vary from operator to operator depending upon his needs, and so can be seen as a software process.

Scrambling

The manner of vision scrambling has already been described in the Vision chapter, but the following is a brief résumé.

For both colour-difference and luminance waveforms, variable cut points are chosen within a determined flight range of 256 equally spaced positions. The positions of the cut points vary from line to line and hence cause the signal to be unintelligible. Two 8-bit numbers, one for each waveform, are required in order to select the cut positions. These are obtained, on each vision line, from a pseudo-random generator.

Sound and data scrambling is achieved by combining an output from a pseudo- random generator with the useful data of each packet.

Both of these scrambling processes are designed to be reversible in the receiver by generating an identical series of numbers to those used in the transmitter. The process of recovering the correct signal

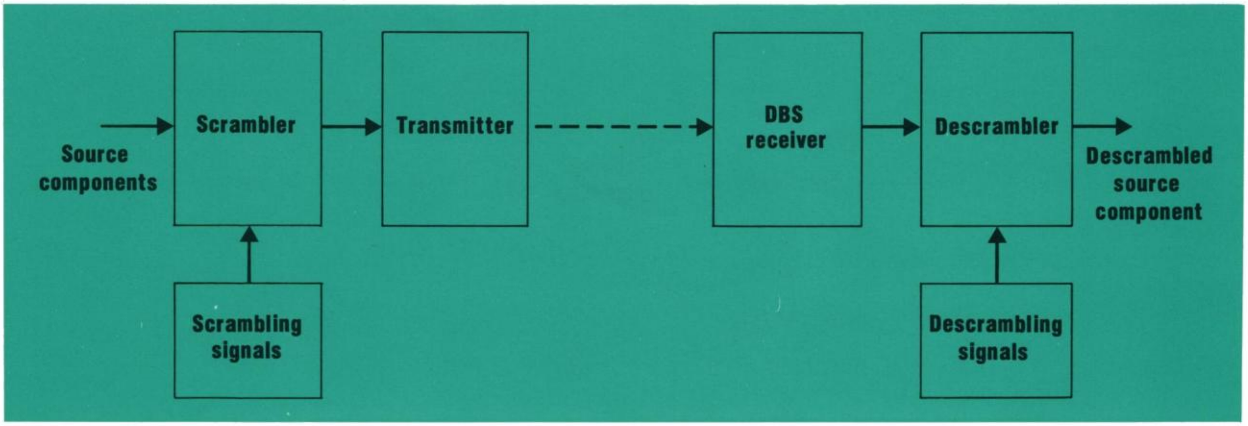


Fig.1. A basic scrambling system.

from a scrambled one is called descrambling. A basic concept of scrambling is illustrated Fig. 1.

Encryption

Locking and unlocking the scrambling process by key signals can be achieved in different ways with varying degrees of sophistication. But whichever process is chosen the unlocking key is always kept secret from the customer. This key, called a Session key, is not used directly by the receiver's descrambling circuits because this would compromise its secrecy. Figure 2 shows the general encryption process.

The encryption system can deliver the Session key to the customer over the air if he is entitled to it; that is, a pay-per-view system. To do this the system must be capable of addressing each customer individually but still offer a high degree of security to the programme provider.

The General Scrambling and Descrambling Process

Figures 3 and 4 illustrate similar scrambling processes but different forms of encryption. In both cases the programme source component indicated can be a picture, sound or data signal which the programme provider may wish to make available to his customers on a conditional basis.

The basic scrambling process is achieved by applying the output of a Scrambling Sequence Generator, which is a continuously changing pseudo-random number, to the source signals. In the receiver the reciprocal process is made possible by re-generating the scrambling sequence in the Descrambling Sequence Generator and applying it in synchronism, along with the scrambled source component, to the descrambler.

In the encoder the Scrambling Sequence Generator's pseudo-random binary sequence has an extremely long cycle time (several million years). The sequence is made even less predictable by using a Control Word (CW) that changes every 256 frames (or approximately every 10 seconds) and a cyclic 8-bit Frame Count (FCNT) that modifies the use of the Control Word every frame. In the receiver the Control Words are provided every 256 frames by a conditional access sub- system which is transmitted in the same satellite channel as the programme source components.

The General Encryption and Decryption Process

Figures 3 and 4 show that to unlock the scrambled signals by using the Descrambling Sequence Generator two of its essential ingredients, the 8-bit Frame Count and Synchronisation, are transmitted in the 'clear'. However, its third ingredient, the Control Word is carried in the conditional access sub-set and can only be recovered by a decryption process based on one of two methods.

Method 1: by regenerating the Control Word in the receiver.

Method 2: by sending the encrypted Control Word to the receiver along with the scrambled programme.

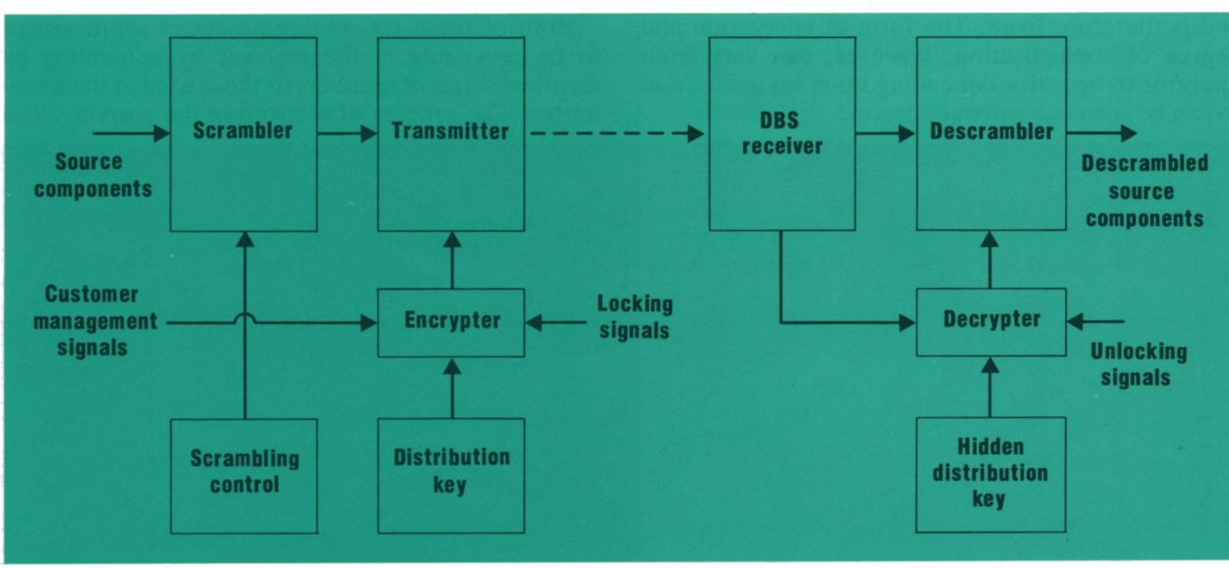


Fig.2. A scrambling system with the addition of an encryption process.

Method 1.
With the use of the Control Word generator situated in the receiver, successful operation depends upon it obtaining the same two conditions as supplied to the transmitter's Control Word generator and shown in Fig. 3. The first condition is the 256 Frame Count (CAFCNT), which for the receiver is obtained from data transmitted in line 625. The second condition is that of the Session Key which, as its name implies, can relate to a type of service and its duration. The receiver obtains this key, in an encrypted form, via a specially transmitted channel called the Customer Management Message. This channel is kept secret from the customer by encrypting it with a Distribution Key which applies only to the user's receiver. Successful decryption of the Session Key is only possible by a properly authorised customer who has the correct Distribution Key. The authorisation can be given user by user, or alternatively it could be a separate key stored with the Control Word generator as a detachable security device.

Method 2.
The alternative method of providing the Control Word is the transmission of an additional channel, the Service Management Message as illustrated in Fig. 4. The Control Word is sent directly to the receiver within the additional channel, but encrypted by the Session Key. At the receiving end the Control Word is then produced by decrypting the Service Management Message with the Session Key. The Session Key, in turn, is carried to the receiver by the Customer Management Message channel in exactly the same manner as described above.

Over-air Addressing
Since in both methods a customer may only access the programmes if his receiver is correctly addressed by the Customer Management Message channel, the term 'Over-air Addressing' equally applies to both techniques of deriving the Control Word.

The choice of the two methods for a particular service will be made by the service operator, as each offers different advantages. In both methods the encrypted signals in the Customer Management Message are always present; but whether or not the customer's receiver is able to make use of them, and regenerate the correct Control Word, depends upon whether the customer has met the required conditions, such as payment. This is the basis of the encryption process.

Pay-per-view
In the second of the two methods, the Customer Management Message channel can also be used to carry the customer's credits or entitlements, shown as (E) in Fig. 5. Carried in the same manner as the Session Key, they are hidden from the customer by encrypting them with the Distribution Key and are then decrypted at the receiver. On receipt they are written into the store of the security device. The contents are then either compared with, or decremented by, the programme-related data (P) which is obtained from the Service Management Message channel. In the case of an impulse pay-per-view, the entitlements (E) represent tokens which are added to the store. The programme data (P) represents the price of the programme which is subtracted from the store when the programme is purchased.

Transmission of the Management Message Channels
A receiver must be informed when a programme is scrambled and where to find the unlocking mechanism within the data multiplex. These details are transmitted in the Service Identification channel in the normal manner (as described in the Service Identification chapter), that is, as a parameter of a Service Command. The parameter, called the Access Related Message (ACCM), informs the decoder if the service contains a Service Management Message and hence whether encryption is present.

The parameter also informs the decoder which method is adopted to produce the Control Word, either by a generator within the receiver or by deriving it from the Service Management Message. If the operator has chosen the Service Management Message method, the parameter will also provide the SMM's packet address together with a pointer to the appropriate subframe which contains the address.

The procedure for determining the packet address of the Customer Management Message channel is similar to above except that the channel is essentially the Over-air Addressing Service. Consequently it can be generally regarded as a service in the same context as the television, radio and teletext services and as such it is described in the Service Identification section. Therefore, as with the other services, it is specified in LISTX (list of indices) whose parameter indicates the packet address of the service's digital component.

In addition to the CMM channel which constitutes the Over-air Addressing Service, a number of CMM channels may exist as service components associated

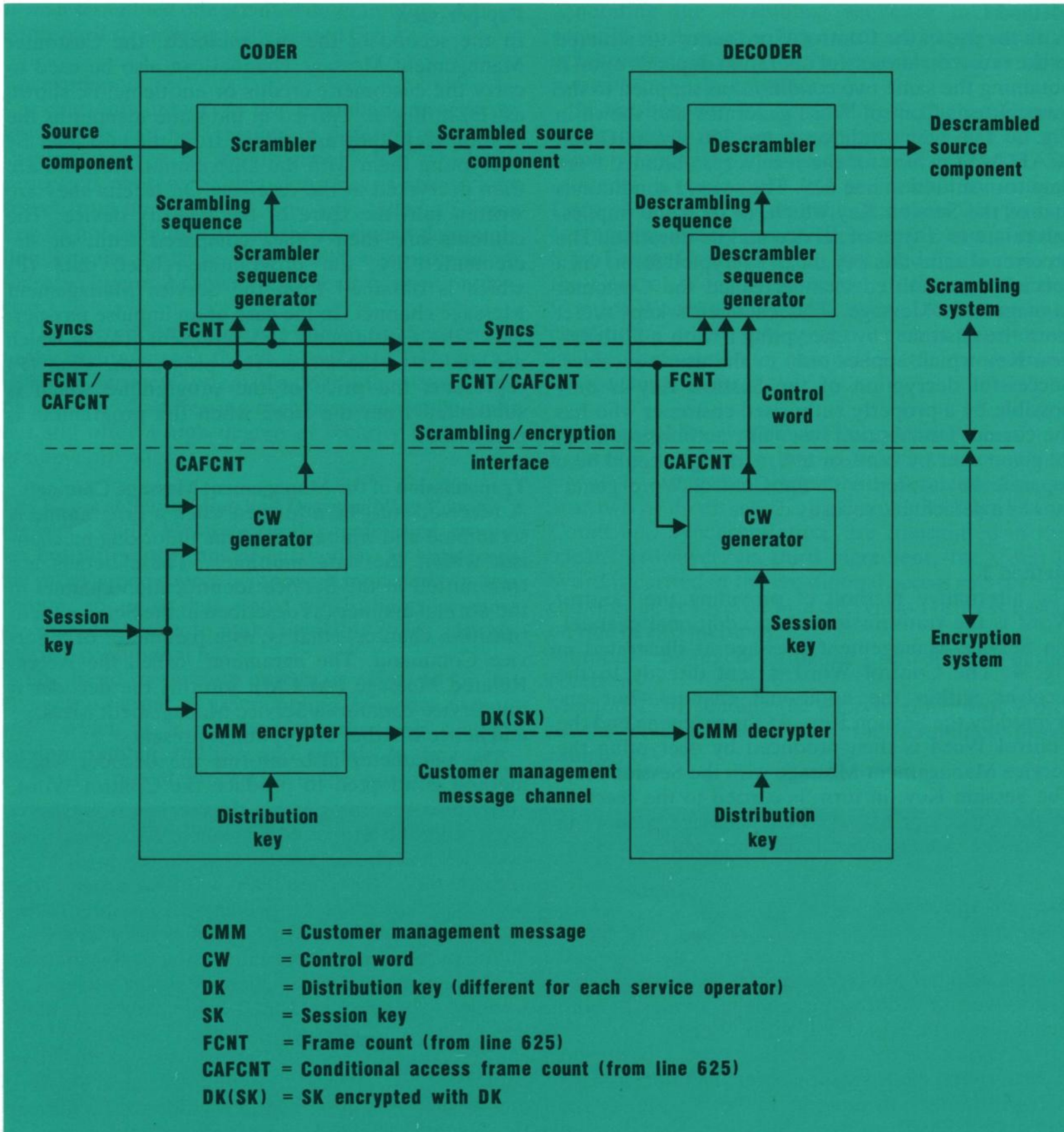


Fig.3. Generalised block diagram of the conditional access system using a Control Word generator (Method 1).

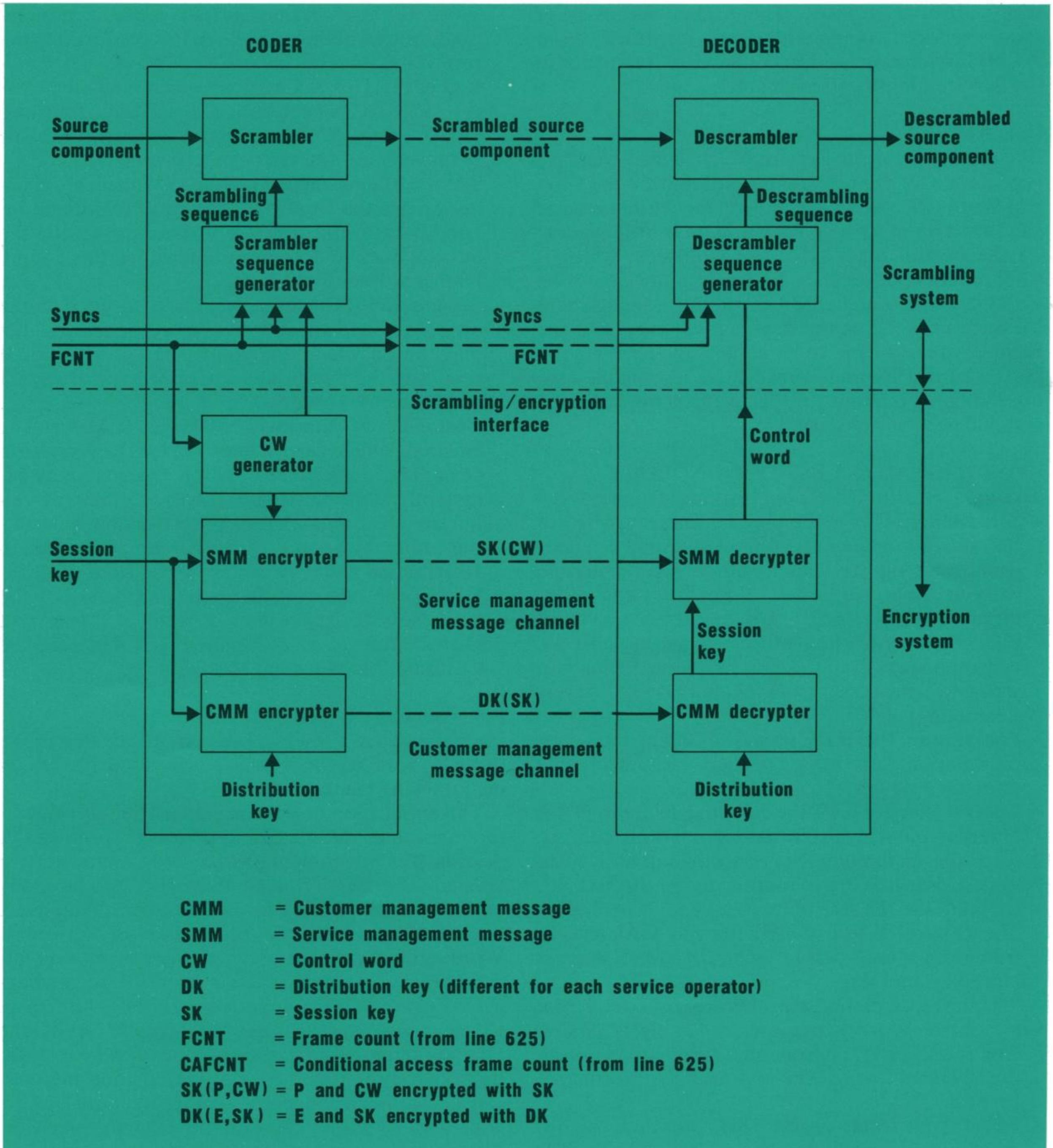


Fig.4. Generalised block diagram showing the Service Management Message method of deriving the Control Word (method 2).

with particular services. The packet addresses of these service components are identified from ACMM parameters within the Service Commands in the Service Identification channel.

Data Rates

The data rate for the Service Management Message is governed by the requirement to transmit the Control Word twice every second. As the entire message can fit within in a single packet there is little loading on the overall capacity of the multiplex, which is 4100 packets per second. An exact figure cannot be given for the number of Customer Management Message packets, as the demand will vary with the number of customers to be addressed and the complexity of distribution. However, a reasonable estimate might be about ten per cent of the capacity, say, 400 packets per second.

CONDITIONAL ACCESS TERMINOLOGY

To summarise the Conditional Access terminology:

- **Scrambling:** This is the process that is applied to the source components of the television, teletext, sound and digital data to make them unusable without descrambling. It is needed allow Conditional Access. (In the MAC/packet system there is another scrambling process which is used for energy dispersal and spectrum shaping; this is a separate process from Conditional Access scrambling).
- **Encryption:** This is the process of hiding the secret information, or keys, needed to unlock the scrambled signals.
- **Control Word:** This is the essential device (a 60-bit binary number) which is used in the transmitter to seed the Scrambler Sequence Generator. The same Control Word is reproduced in the receiver to seed the Descrambler Sequence Generator. The Control Word is used for 256 MAC/packet frames and then, before being discarded, it is replaced by another.

  In the receiver the Control Word is obtained from a Control Word generator seeded by a Session Key (method 1), or from an encrypted version of the Control Word via a transmission channel (method 2).
- **FCNT 8-bit frame count:** This a continuously incrementing count from 0 to 256 MAC/packet frames. At a count of 255 the next frame will cause it to be reset to zero and CAFCNT to be incremented. It is used in the scrambler and descrambler circuits to modify the use of the Control Word for each frame and so vary the pseudorandom scrambling sequence. It is conveyed to the receiver in line 625.
- **CAFCNT:** This is a number which increments every 256 MAC packet/frames. It seeds the transmitter's Control Word generator when no Service Management Message channel is used (method 1).
- **Service Management Message (SMM):** This channel carries the Control Word and Programme related data to the receiver. It is encrypted by the Session Key and transmitted within the Sound/Data multiplex as packets.
- **Session Key:** The basic key used to encrypt the SMM, or seed the Control Word generator, depending on which encryption method is used. It relates to the duration of a service, type of service and to the user group.
- **Customer Management Message (CMM):** The channel which carries the Session Key and the customers' entitlements to the receiver, it is encrypted by the Distribution Key. Commonly called the Over-air Addressing service and transmitted within the Sound/Data multiplex as packets.
- **Distribution Key:** A key kept secret from the user. Within the decoder successful production of the basic Session Key will only result if the correct Distribution Key is applied to the received Customer Management Message.

Encryption in a Layered Pattern

From the above and Figs. 3-4 it canbe seen that in the receiver both methods follow a layered pattern in the following manner:

The application of the correct Distribution Key to the Customer Management Message produces a Session Key which contains the programme service details. The Session Key may also produce any entitlements applicable to the customer. Obtaining a Session Key allows the production of a Control Word either by means of a generator or by decrypting it from the Service Management Message channel. A Control Word together with the 8-bit Frame Count and synchronisation signals seed the Descrambling Sequence Generator which, in turn, unlocks the programmes. Both encryption methods can be further refined by adding additional layers and extra features; for example, the pay-per-view feature or a method of providing shared groups.

A SHARED KEY OVER-AIR ADDRESSING SYSTEM

Having described the general case of conditional

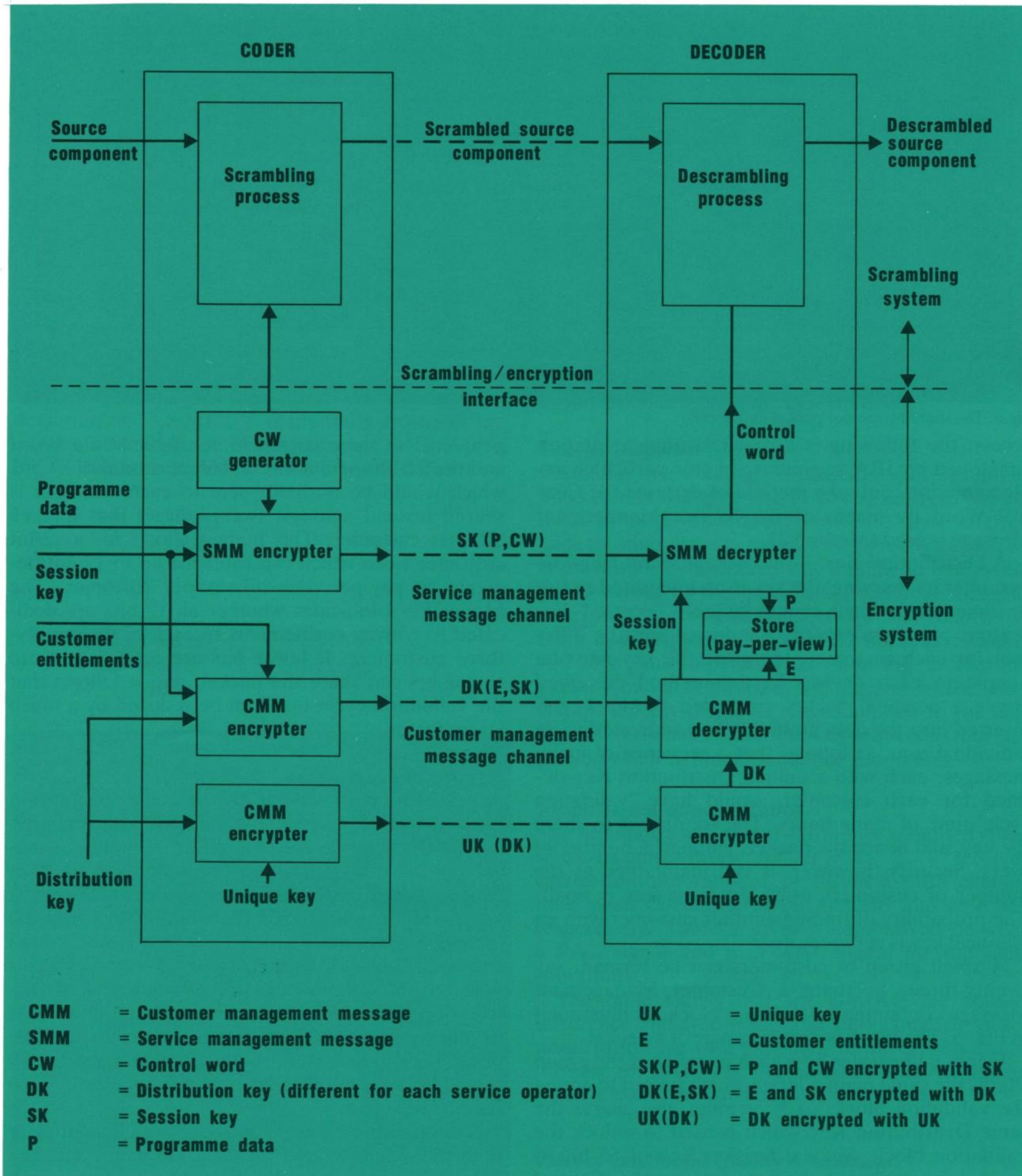


Fig.5. Encryption using the Service Management Message method with the addition of pay-per-view and a shared distribution key.

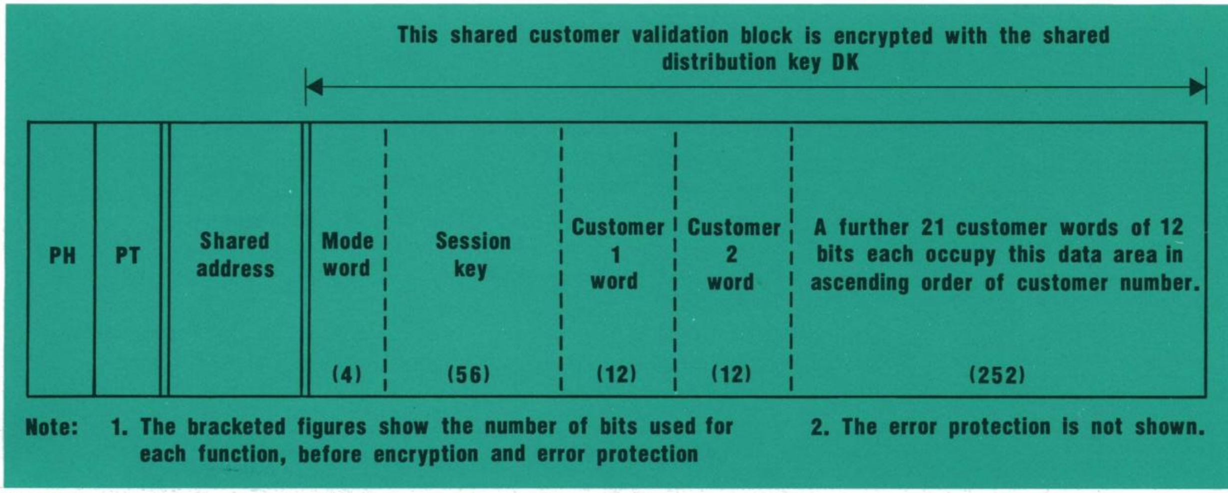


Fig.6. The shared customer packet data format.

access, the following is a practical implementation developed by IBA engineers. In this particular implementation, only the method of deriving the Control Word by means of the Service Management Message is used (Method 2).

A Distribution Key is always kept secret from the customer by 'burying' the key in an integrated circuit or some device which cannot be easily accessed. For reasons of system security the device must be different for each customer, therefore the key must be unique to each device. In the method described above a personal packet encrypted by the key is inserted into the data multiplex and addressed to an individual user. It follows that a sequence of many messages, each with a unique Distribution Key defined for each customer, would have 'validation cycle' time of many hours. The solution is to share the key, and hence the packet, amongst a number of users. Security, however, is still maintained as the number of customers using the same key is small. The probability of finding another customer with an identical key is very remote.

A small group of customers can be formed, say twenty-three, to share a Customer Management Message containing a validation block, as illustrated in Fig. 6. The block is shown as part of a packet.

Figure 6 illustrates how each member of the group shares the same main address which is used to access the validation block; each member also shares the same Distribution Key which is used to unlock the validation block. Since a Session Key of 56 bits is needed by each member to recover the Control Word and hence descramble the programme, its 'overhead' in bits is shared amongst twenty-three people. The same is true of the 4-bit Mode Word and the 24-bit main address. Hence a total of 84 bits which would be normally sent to each customer is shared instead amongst twenty-three; that is, 3.65 bits per customer. This budget allows for a 12-bit customer word which can cater for up to 12 entitlements or pay-per-view tokens per customer. The Mode Word indicates whether all 12 bits are dedicated to convey entitlements to each of the twenty-three customers. If fewer bits are used then more customers can share the packet, and it follows that the validation cycle time can be reduced by a much larger factor.

Pirating of Programmes

A problem could arise if one customer becomes a pirate, in which case the obvious solution of removing his Distribution Key will affect all other customers sharing that same key. The remedy is to store two keys in a customer's security device. The first key is the normal shared Distribution Key and the second is a Unique Key (UK) which is not shared and is different for each customer. When a pirate is detected the operator constructs a new shared Distribution Key which is sent to the honest customers by encrypting it with their Unique Key, thus the pirate is excluded. Figure 5 illustrates this concept by the addition of another layer containing the Unique Key arrangement.

Although these unique messages are less efficient than the single Shared Distribution Key cycle, the unique customer cycle includes only a very small number of customers. The complete process takes less than one minute.

RF Transmission of D-MAC/ packet

Synopsis
This chapter describes the performance of a D-MAC/ packet signal in a satellite channel. This is illustrated in tabular form with the addition of graphs showing the subjective results of picture quality under varying carrier-to-noise ratios. Comparison is also made between the performance of sound signals in a 27 MHz channel and in a 21 MHz channel. In the former case NICAM and Hamming parity codings are used incorporating the conventional decoding techniques, while in the latter case Viterbi sound decoding is shown to allow improved reception of lower carrier-to-noise ratios.

Modulation and Pre-emphasis

For satellite transmission the complete baseband D-MAC/packet signal is frequency modulated with a deviation of 13.5 MHz/V at the 0 dB crossover frequency of the pre-emphasis network, as shown in Fig. 1. The pre-emphasis characteristic is defined by:

$$H(f) = A \times \frac{1+j\,(f/f_1)}{1+j\,(f/f_2)}$$

where $A = 1\sqrt{2}$

$f_1 = 0.84$ MHz

$f_2 = 1.5$ MHz

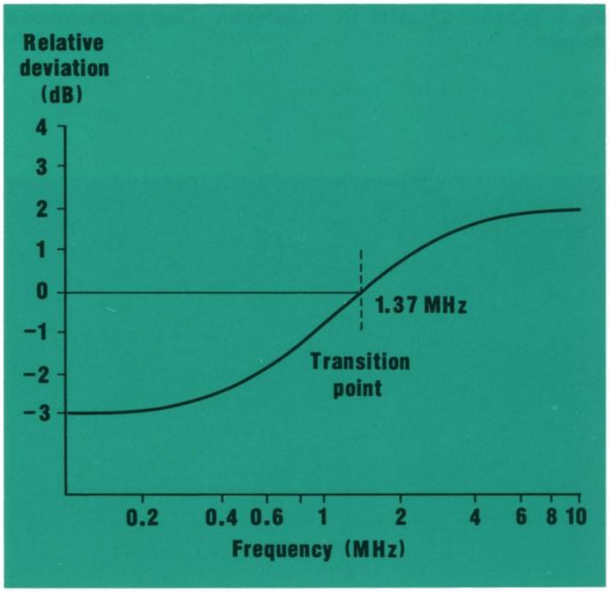


Fig.1. MAC pre-emphasis network.

A comparison with CCIR Recommendation 405-1 for PAL shows that considerably more low frequency deviation is used in the case of MAC.

Energy Dispersal

Particular modulating conditions (for example, black level) will give rise to strong discrete components in the FM spectrum and could cause interference to other users in the same frequency band. This is prevented by arithmetically adding an energy dispersal signal to the modulating vision signal. The energy dispersal signal consists of a triangular waveform at a frequency of 25 Hz having a deviation of 600 kHz peak-to-peak as shown in Fig. 2. This dispersal signal is removed in the receiver by the video clamp. Energy dispersal is also applied to the data signals by means of a scrambling sequence (that is, a pseudo-random binary sequence generator).

Vision Signal-to-noise Ratios

The luminance and colour-difference weighted video signal-to-noise ratios for D-MAC at a received carrier-to-noise ratio of 14 dB in a DBS channel have been calculated to be 42.5 dB and 43.6 dB, respectively. Table 1 gives a comparison between a PAL signal and a D-MAC signal for the same conditions.

The table shows that for D-MAC the colour-difference signal-to-noise and luminance signal-to-noise ratios are well matched, however, this is not the case for PAL. The performance of MAC, unlike PAL, is not dominated by poor colour noise.

Picture and Sound Quality Assessment

To determine the level of MAC picture quality under varying carrier-to- noise ratios a number of subjective tests have been conducted using test

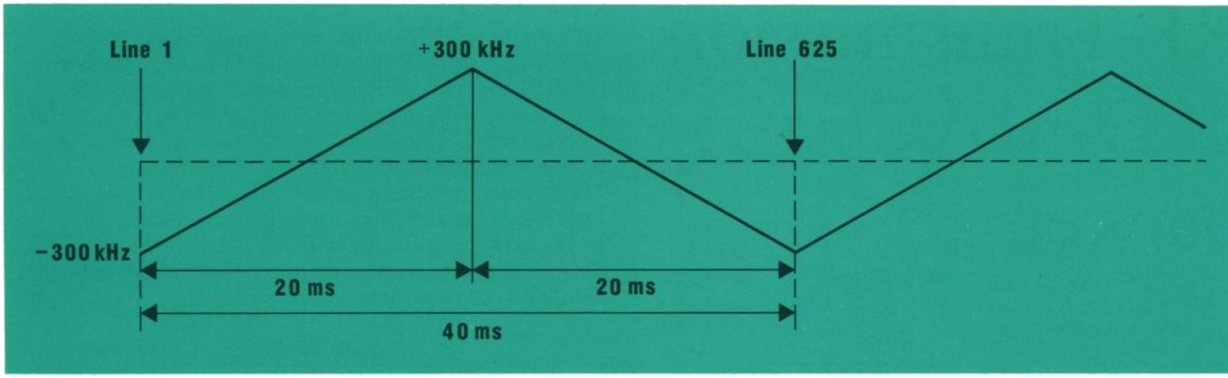


Fig.2. Energy dispersal added to the baseband signal.

Signal	Video S/N ratio, dB	
	PAL	MAC
Luminance	44.1	42.5
Chrominance U	40.9	46.6
Chrominance V	43.9	46.6
Combined U and V	39.2	43.6

Table 1 Comparison between PAL and D-MAC video signal-to-noise ratios for the same conditions.

slides. During the tests a panel of expert observers rated the picture quality according to the CCIR Recommendation 500-1 five-point quality scale:

5=excellent
4=good
3=fair
2=poor
1=bad

The test transmissions were conducted with two bandwidths, 27 MHz (the full WARC bandwidth) and 21MHz. The results showing D-MAC quality vs. carrier-to- noise ratio were made using a conventional (that is a non threshold-extension) demodulator and are given in Figs. 3 and 4. It can be seen that good quality vision is obtained with a carrier-to-noise ratio of 12 dB.

Also shown is the subjective quality of the sound signals using NICAM and Hamming parity coding. These tests were conducted using the same bandwidths as for the vision, but with two sound decoders. In the case of the 27 MHz channel a conventional sound decoder was used, whereas results in the case of the 21 MHz channel were obtained in conjunction with a Viterbi sound decoder.

From the above it was found that reception of a 27 MHz channel bandwidth with a conventional sound

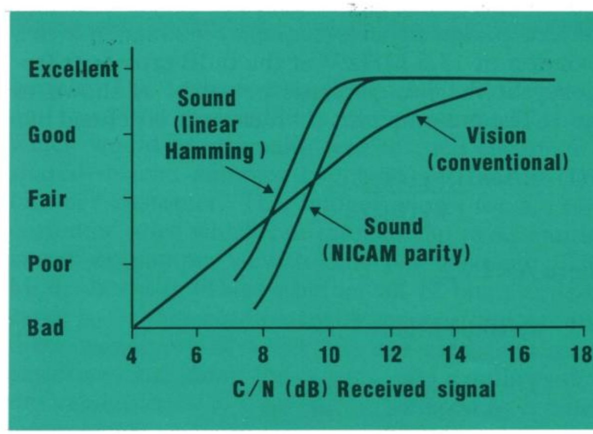


Fig.3. D-MAC 27 MHz I.F. bandwidth using a conventional decoder.

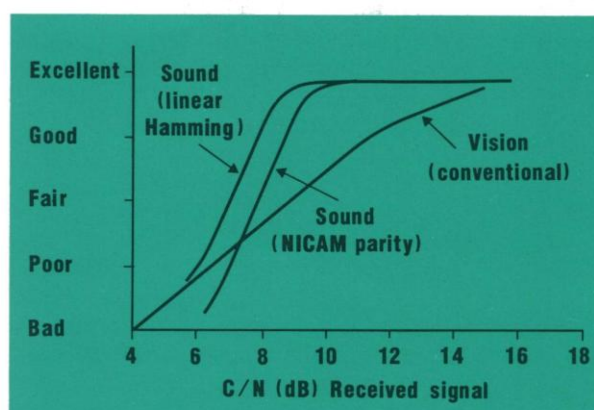


Fig.4. D-MAC 21 MHz I.F. bandwidth using a Viterbi decoder.

decoder will provide good quality sound at a carrier-to-noise ratio of about 10.5 dB if Nicam parity coding is used. However, if linear Hamming coding is used with a 21 MHz bandwidth channel in conjunction with Viterbi decoding, the sound quality is still good at the much lower carrier-to-noise ratio of 8 dB.

Reception of Essential Data

Equally important is the requirement to correctly decode the Service Information channel. This data, which provides the receiver with a description of the satellite network and its individual services, must operate successfully under low signal conditions. In the instance of a fade, caused by bad weather, it would be undesirable for the receiver to 'hop' from service to service as a result of poor decoding. Table 2 shows the specified performance of the Service Information channel under varying conditions. The figures are based upon an IF bandwidth of 21 MHz and a Viterbi decoder; to date measured results are not available.

Carrier/noise ratio, dB:	10.5	8	4
BER:	10^{-5}	10^{-3}	3×10^{-2}
Access to main sound component	0.5	1	2
Access to a complete service	0.75	2	5

Table 2. Specified performance of Service Identification acquisition. (Access time in seconds).

It can be seen that the D-MAC/packet signal is well matched to the DBS transmission channel, offering a good balance between the signal-to-noise ratios of the colour-difference and luminance components.

A basic receiver will have an FM threshold and digital sound threshold (bit- error-ratio of 1 in 10^{-3}) in the region of 10-11dB. A more sophisticated receiver using Viterbi techniques will allow sound decoding at carrier-to-noise ratios of 8-9dB.

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Satellitensender f ür das vereinigte königreich

von M D Windram, G J Tonge und R C Hills

Zusammenfassung

Das Vereinigte Königreich steht gerade an der Schwelle einer neuen und interessanten Phase in der Geschichte seines Rundfunks. Im August 1989 startet nämlich der erste Satellit für die Satelliten-Direktsendung, der ausschließlich von der privaten Hand finanziert wird. Um den kommerziellen Erfolg für diesen Service abzusichern, muß jedoch der Anbieter schnell einen großen Markt erreichen und gleichzeitig das Potential für verbesserte Empfangsgeräte maximal nutzen, um das Marktinteresse wach zu halten. Die Wahl des D-MAC/Paket-Systems als Signalformat für diesen Service ist gegen diesen Hintergrund zu sehen. Das System bietet eine Empfangsoption, bei der ein relativ preisgünstiger, auf das Empfangsgerät passender Signalkonverter zur Anwendung kommt, so daß schnelle Marktdurchdringung gesichert ist. D-MAC bietet jedoch eine Reihe von weiteren Service-Optionen, die anhaltendes Marktinteresse stimulieren dürften. Für den Bildteil des Systems sind für den Empfänger vier Phasen vorgesehen: Vorhandener Empfänger mit Konvertergerät, integriertes MAC-Empfangsgerät, Breitbild-Empfangsgerät sowie HDTV-Empfangsgerät. Der Digital-Teil des Signals bietet eine Vielzahl von Tonübertragungsmöglichkeiten sowie Möglichkeiten für die Datenübertragung, ein Bereich mit wachsendem Interesse. Diese stellen weitere Einnahmequellen für den Systembetreiber dar, wobei HDTV einen Steuerkanal für die wesentlich verbesserte Dekodierung im Emfangsteil bietet.

Wenn man von den Empfangsgeräten und dem Übertragungssystem absieht, sind zusätzliche Studios und Programmquellen für die Bedienung der vorhandenen und der neuen Systeme notwendig. Im Vereinigten Königreich soll der ideale HDTV-Studiostandard 1250 Zeilen bei einer Austastfrequenz von 50 Hz aufweisen. Dies paßt gut zu den gegenwärtigen Film- und Videostandards und eignet sich hervorragend als Quelle für ein MAC-Kodiersystem mit hoher Auflösung für die HDTV-übertragung.

Eine übersicht über das D-MAC/Paket-System

Zusammenfassung

Dieses Kapitel soll eine Übersicht über das Gesamtkonzept des D-MAC-Übertragungsstandards vermitteln. Hier wird die Art der Multiplex-Analogbausteine behandelt. Es folgt eine Überleitung in eine Beschreibung der Wellenform für den Videobereich, der Gründe für die Zeitraffung und die gewählten Signalabtastungswege. Vor der Betrachtung der Ton- und Datenkanäle wird die Wahl der duobinären Kodierung sowie der Aufbau der Paketstrukter betrachtet. Die Grundlagen der Verwürfelungs- und Verschlüsselungsverfahren werden erörtert, sowie die Kabelanforderungen für D und D2-MAC.

Es wird davon ausgegangen, daß der Leser nach dem Studium dieses Kapitels den Inhalt eines beliebigen einzelnen Kapitels verarbeiten kann, also nicht mehr einzelne Kapitel in einer vorgegebenen Reihenfolge betrachten muß.

Das Bildsignal

Zusammenfassung

In diesem Kapitel wird ein Vergleich zwischen der Struktur von D-MAC und PAL vorgenommen. Das Format der Wellenform im D-MAC-Grundband wird zusammen mit dem Aufbau eines kompletten 625-Zeilen-Bilds beschrieben. Es folgen Gründe für die Anwendung der Bildzeitraffung, und eine Erörterung der Konsequenzen für die Bandbreite sowie der Sequentialübertragung des Farbdifferenz-Signals. Dieses Kapitel befaßt sich auch mit der Synchronisierung, Pegelhaltung, Bildsignalverwürfelung und den Maßnahmen für die Übertragung von Breitbildformaten.

Paketkonstruktion und übertragung

Zusammenfassung

Hier werden die Hintergründe für die möglichen Ton/Daten-Kodierformate vor der Einführung des D-MAC/Paketformats betrachtet. Vor der Beschreibung der D-MAC/Paketstruktur werden in diesem Kapitel die Grundverfahren für die duobinäre Kodierung und die Möglichkeiten zur Steigerung der Bit-Kapazität betrachtet. Die Struktur eines individuellen Pakets einschließlich der Funktionen des Paketvorsatzes und der gelegentlichen Notwendigkeit einer 8-Bit-Pakettype wird beschrieben.

Die flexible Art der Multiplex-Übertragung wird anhand von Abbildungen betrachtet, mit denen gezeigt wird, wie ein vollständiges, aus zwei Ton-/Daten-Teilbildern zusammengesetztes Bild für 4100 Pakete/Sekunde und eine mittlere Bitzahl von 2,952 Mbit/s ausreicht. Außerdem werden die Alternativmöglichkeiten für die beiden Teilbilder zusammen mit den Anforderungen für die Paketübertragung erörtert.

Tonkanäle

Zusammenfassung

Die Kapazität des D-MAC Ton-/Datenkanals reicht für bis zu acht Tonkanäle von hoher Qualität oder bis sechzehn Stimmkanäle aus. Zusätzlich können Daten mit einer Kapazität übertragen weden, die weit über der von vorhandenen, erdgebundenen Kanälen liegt. Für jeden Qualitätskanal sind ein oder zwei Fehlerschutzstufen möglich, und der Verlauf kann linear oder NICAM-kodiert sein. Dieses Kapitel enthält die Beschreibung der vorhandenen Möglichkeiten unter vereinfachter Betrachtungsweise der NICAM-Kodieroption.

Gleichzeitig wird die Einbeziehung der verschiedenen Tonkanäle in die Paketstruktur zusammen mit den Gesamtanforderungen für die Ton- und Daten-Multiplexübertragung beschrieben.

Auswertungsblöcke (BI)

Zusammenfassung

Die Information in den sog. Auswertungsblöcken gestattet dem Tondekoder die automatische Einstellung des Audiopegels für die gewählte Betriebsart und bereitet den Dekoder gleichzeitig auf zukünftige Änderung im angewählten Service vor. Diese Information, die in Form von Datengruppen vorliegt, wird über Pakete im Ton-/Daten-Multiplexverbund übertragen. Die Zusammensetzung der Befehle und deren Organisation in Datengruppen wird zusammen mit einem Beispiel für die Funktionsweise der Auswertungsblöcke beschrieben.

Service-Identifikationssignale

Zusammenfassung

Angaben über den Satelliten-Sendekanal sowie über das verfügbare Bild-, Ton- und Datenangebot werden über den Service-Identifikationskanal zum Empfänger übertragen. Dieser umfassende Informationsbestand ist in verschiedenen Stufen, angefangen vom einfachen Parameter eines Befehls bis zum komplizierten Format einer Datengruppe, angeordnet. Wie bei anderen Daten im Ton-/Daten-Multiplexverbund wird auch diese Information in Paketen übertragen. Dieses Kapitel verfolgt den Weg eines Parameters durch die einzelnen Schichten der Struktur und enthält ein Beschreibung jeder Stufe. Am Ende des Kapitels findet sich eine Beschreibung eines typischen Satelliten-kanals und die erforderliche Kapazität für die Beschreibung des Service-Angebots.

Datensendungen

Zusammenfassung

Der Ton-/Datenimpuls im D-MAC-Paketsystem kann so konfiguriert werden, daß nicht nur Ton und Daten im erforderlichen Umfang für das Management des Multiplex-Verbunds übertragen werden, sondern auch Videotext, Untertitel, Standbilder und transparente Datenkanäle und vieles andere. Das System ist somit außergewöhnlich flexibel und kann noch flexibler gestaltet werden, indem das gesamte Feld für die Datenübertragung genutzt wird, wenn das Bildsignal nicht gebraucht wird.

Dieses Kapitel befaßt sich hauptsächlich mit dem Aufbau der beiden Videotext-Datenblöcke. Beide leiten sich aus der CCIR-Empfehlung 653 System B (Weltsystem Videotext) ab, unterscheiden sich jedoch in der Auslegung der Schutzart. Das Kapitel enthält eine Beschreibung der Schutzstufen und der Art und Weise der Einfügung von Videotext-Paketen in ein D-MAC-Paket mit Einzelheiten über die Signalführung im Service-Identifikationskanal.

Prioritätsmeldungen - Zeile 625

Zusammenfassung

Die Hauptfunktion dieser Zeile, die vollständig aus Daten besteht, ist die Organisation der Struktur des gesamten Multiplex-Übertragungsverbunds. Die Daten dieser Zeile gehören in zwei Hauptkategorien. Erstens handelt es sich um statische Information, die zwar für den Empfänger wichtig ist, aber von einem Bild zum nächsten unverändert bleibt, zweitens um dynamische Information, die sich immer wieder ändert und aus diesem Grunde bei jedem Bild neu zum Empfänger übertragen werden muß. Beispiele für Informationen, die im statischen Datenblock zu übertragen sind, sind die Kennung der Satelliten-Position, sowie Datum und Uhrzeit. Dynamische Informationen beschreiben die Multiplex-Konfiguration, die sich je nach den Anforderungen des Betreibers von Zeit zu Zeit ändert.

Abschließend folgt eine Liste der verschiedenen Datenblöcke in den beiden Kategorien zusammen mit einer Beschreibung der jeweiligen Funktionen.

Bedingter Systemzugriff

Zusammenfassung

Unter Verwendung eines als Verwürfelung bezeichneten Verfahrens kann das vom Satelliten übertragene Programm für Unbefugte unbenutzbar gemacht werden. Eine Verwürfelung reicht jedoch dann nicht aus, wenn der Betreiber des Kanals ein Format wünscht, bei dem Zahlung beispielsweise nach Zeit erfolgt. Die Signale können verwürfelt sein, jedoch ist ein Gerät erforderlich, daß die Zerwürfelung aus- und wieder einschaltet. In diesem Kapitel werden beide Prozesse beschrieben, sowie die Art und Weise der Übertragung des "Schlüssels" zum Empfänger über einen getrennten Kanal. Anschließend folgt eine Erörterung eines Zahlungssystems nach Zeit mit Beschreibung der praktischen Realisierung eines gemeinsamen Schlüssel-Adressiersystems und Vorschlägen für die Verhinderung des unbefugten Kopierens von Programmen. Außerdem werden die Anforderungen an die Übertragungs- und Datenkapazität betrachtet.

HF-Übertragung des D-MAC/Pakets

Zusammenfassung

Dieses Kapitel beschreibt die Verhaltensweise eines D-MAC/Paketsignals in einem Satellitenkanal. Zur Erläuterung wird ein Tabelle verwendet, die durch Diagramme ergänzt wird, aus denen die subjektiven Ergebnisse in bezug auf Bildqualität bei verschiedenen Träger-Rauschabständen hervorgehen. Außerdem wird ein Vergleich zwischen der Verhaltensweise von Tonsignalen auf einem 27 MHz-Kanal und einem 21 MHz-Kanal angestellt. Im vorigen Falle werden NICAM- und Hamming-Paritätskodes verwendet, bei denen konventionelle Dekodierverfahren angewandt werden. Im letzteren Falle wird gezeigt, daß die Viterbi-Tondekodierung verbesserten Empfang bei niedrigeren Träger-Rauschabtsänden zuläßt.

Radiodiffusion par satellite dans le Royaume Uni

par M D Windram, G J Tonge, R C Hills

Résumé

Le Royaume Uni va entrer dans une phase nouvelle et passionnante de l'histoire de sa radiodiffusion. En Août 1989 le satellite pour le premier service haute puissance de radiodiffusion directe par satellite (DBS) sera lançé et il est financé entièrement par des particuliers. Pour garantir le succès commercial de ce service, l'émetteur doit établir rapidement un marché important mais aussi offrir le potentiel d'améliorations du récepteur pour conserver l'intérêt du marché. Sur ce plan là, le système/D-MAC a été adopté comme format signal de ce service. Ce système offre une option réception impliquant un adapteur externe et relativement peu cher et qui devrait donc favoriser une entrée rapide dans le marché. Néanmoins, D-MAC offre aussi un certain nombre d'options de service qui devraient stimuler l'intérêt continu du marché. En ce qui concerne la partie vision du système, quatre phases sont décrites pour le récepteur: le récepteur existant avec l'adapteur externe, le récepteur intégré MAC, le récepteur grand écran et le récepteur HDTV. La partie données numériques due signal offre une variété d'options son de haute qualité ainsi que des services de données qui représentent un domaine à l'intérêt croissant. Elles donnent aussi l'occasion de générer des revenus pour l'émetteur et dans le contexte du HDTV offrent un canal de contrôle qui permet d'effectuer un meilleur décodage du récepteur. En travaillant à partir des récepteurs et du système de transmission, on a besoin de studios et de sources qui alimenteront les services nouveaux et déjà existants. Dans le cadre du Royaume Uni, le standard studio HDTV idéal utilisera 1250 lignes et un taux de zone de 50-Hz. Cela s'accorde bien avec le cadre vidéo et film déjà existant et est idéal comme source d'un système de codage haute définition MAC pour la transmission HDTV.

Un aperçu du système D-MAC/Paquet

Résumé

Une appréciation due concept général du standard de transmission D-MAC est donnée dans ce chapitre. La nature des Composantes Analogues de Multiplexage (MAC) est couverte et amène à une description de la forme d'onde de vision, des raisons des compressions de temps et des taux d'échantillonnage choisis. Avant de considérer les détails des canaux de données et de son, on s'arrête aussi sur le choix du codage duobinaire et sur la formation de la structure du paquet. La base des techniques de brouillage et de chiffrage est mentionnée, suivie par les spécifications du cable pour D et D2-MAC.

On espère qu'après avoir lu ce chapitre, le lecteur pourra apprécier le contenu de tous les chapitres individuels et n'aura pas forcément besoin de les aborder dans un ordre strict.

Le signal d'image

Résumé

Dans ce chapitre on établit une comparaison entre la structure de D-MAC et de PAL. Le format de la forme d'onde D-MAC de la bande de base est décrit ainsi que la composition de la trame complète de 625 lignes. On justifie l'utilisation de la compression due temps de vision et les conséquences sur la largeur de bande sont débattues ainsi que la transmission séquentielle des signaux de différence – couleur. Ce chapitre comprend aussi des sections concernant la synchronisation, la référence de niveau noir, le brouillage de vision et les arrangements pour la transmission d'images plus larges par rapport à la hauteur.

Construction et transmission paquet

Résumé

Le contexte des formats de codage son/données possibles est examiné avant l'adoption due format D-MAC/paquet. Avant que la structure du D-MAC/paquet ne soit décrite, le chapitre s'intéresse aux techniques de base du codage duobinaire et à la façon par laquelle on peut réussir une augmentation de la capacité de bits. La structure d'un paquet individuel est décrite y compris les fonctions de l'en – tête du paquet avec le besoin occasionnel du type de paquet 8-bits.

La nature souple du multiplex est examinée accompagnée d'illustrations pour montrer comment une trame entière de deux sous-trames son/données peut supporter 4100 paquets par seconde donnant un taux net de bits moyen de 2.952 Mbits/s. Les autres usages de ces deux sous-trames sont aussi examinés ainsi que les caractéristiques de transmission de paquet.

Canaux son

Résumé

La capacité du canal son/données D-MAC permet jusqu'à huit canaux son de haute qualité ou seize canaux commentaires, tous étant accompagnés de données avec une capacité dépassant un canal terrestre existant. Chaque canal de haute qualité peut avoir un ou deux niveaux de protection d'erreur et peut être codé linéairement ou NICAM. Ce chapitre offre une description de ces possibilités avec un traitement simplifié de l'option de codage NICAM.

L'insertion des différents canaux son dans la structure du pacquet est décrite avec l'ensemble des caractéristiques multiplexes son/données.

Blocs de decodage (BI)

Résumé

Les informations contenues dans les Blocs de Décodage permettent au décodeur de son de fournir automatiquement la sortie son correcte pour le service choisi et prépare le décodeur à tout changement futur dans le service choisi. Cette information, sous forme de groupes de données, est transmise par paquets dans le multiplex son/données. La composition des commandes et leur organisation en groupes de données est décrite avec un exemple de l'opération Bloc de Décodage.

Signaux d'identification de service

Résumé

Les informations concernant le canal de transmission par satellite ainsi que les détails des services de l'image, du son et des données disponibles, sont transmis au récepteur au moyen du canal d'Identification de Service. Cette grande quantité d'informations est assemblée de façon superposée allant du paramètre de base d'une commande au format complexe d'un Groupe de Données. Comme avec d'autres données dans le multiplex son/données, cette information est transmise par paquets. Ce chapitre suit le chemin d'un paramètre à travers les couches de la structure tout en fournissant une description de chaque étape. Un exemple de canal satellite typique et la capacité requise pour décrire ses services est fourni à la fin du chapitre.

Radiodiffusion de données

Résumé

Le groupe son/données du système de paquet D-MAC peut être configuré pour transmettre non seulement le son et les données requis pour diriger le multiplex complet, mais aussi pour passer du teletext, des sous-titrages, des images immobiles et des

canaux de données transparents et plus encore. Le système est très souple et peut l'être encore plus en utilisant le champs complet pour la transmission de données quand le signal de vision n'est pas nécessaire.

Le contenu de ce chapitre concerne essentiellement la construction de deux formes de bloc de données teletext. Elles dérivent toutes les deux du système B 653 de Recommendation CCIR (World System Teletext) mais diffèrent par leurs niveaux de protection. Une description des niveaux de protection et comment un paquet de teletext complet est inséré dans un paquet D-MAC avec signalisation dans le canal identification de Service est fourni.

Informations prioritaires – Ligne 625

Résumé

La fonction principale de cette ligne, faite entièrement de données, est d'organiser la structure de tout le multiplex de transmission. Les données dans la ligne se divisent en deux catégories. Premièrement l'information statique qui tout en étant importante pour le récepteur est en fait constante de trame à trame. Deuxièmement, l'information dynamique qui change et donc qui doit être rapidement transmise au récepteur trame par trame. Des exemples d'information transmise par le bloc de données statique comprennent l'identification de la position du satellite ainsi que la date et l'heure. L'information dynamique décrit la configuration du miltiplex qui peut changer de temps en temps selon ce que demande l'émetteur.

Une liste des divers blocs de données dans les deux catégories est fournie, ainsi qu'une description de leurs fonctions.

Le systeme d'acces conditionnel

Résumé

Les transmissions par satellite peuvent être rendues inintelligibles à l'utilisateur par un procédé appelé communément brouillage. Néanmoins, le brouillage par lui-même n'est pas suffisant si l'opérateur du canal désire un format à prépaiement. Les signaux eux-mêmes peuvent être brouillés mais un dispositif est nécessaire pour verrouiller et déverrouiller le procédé de brouillage, ceci est appelé chiffrage. Le chapitre décrit ces deux procédés et la façon dont on peut passer les 'codes' de déverrouillage au récepteur par des canaux de transmission séparés. Un système à prépaiement est examiné, suivi par une mise en application pratique d'un système de partage de code par satellite, ainsi qu'une suggestion pour éliminer le piratage des programmes. Les exigences de transmission et de capacité de données sont aussi examinées.

Transmission haute frequence de D-MAC/Paquet

Résumé

Ce chapitre décrit la performance d'un signal D-MAC/paquet dans un canal à satellite. Elle est illustrée sous forme de tableaux et de graphiques montrant les résultats subjectifs de qualité d'image sous des rapports porteuse/bruit divers. Une comparaison est aussi établie entre la performance des signaux son dans un canal de 27 MHz et dans un canal de 21 MHz. Dans le premier cas, les codages de parité NICAM et Hamming sont utilisés en incorporant les techniques de décodages conventionnelles, tandis que dans le dernier cas il est démontré que le décodage son Viterbi permet d'améliorer la réception de rapports porteuse/bruit inférieurs.

Radiodifusion via satelite en el Reino Unido

por M D Windram, G J Tonge, R C Hills

Resumen

El Reino Unido está a punto de entrar en una nueva y excitante fase de su historia de radiodifusión. En agosto de 1989 se lanzará el satélite para el primer servicio de DBS (Radiodifusión Directa por Satélite) de alta potencia del mundo, que está financiado privadamente en su totalidad. Para el éxito comercial de este servicio el radiodifusor necesita establecer un amplio mercado rápidamente y sin embargo necesita ofrecer el potencial de mejoras de recepción para mantener el interés del mercado. Teniendo esto en cuenta, ha sido adoptado el sistema de paquete D-MAC como el formato de señal para este servicio. Este sistema ofrece una opción de recepción que incluye un convertidor adosable y por tanto permitirá una rápida penetración en el mercado. Sin embargo, D-MAC también proporciona un número de opciones de servicio que estimulará el continuo interés del mercado. Para la parte de visión del sistema se describen cuatro fases para el receptor: el receptor existente con convertidor adosable, el receptor MAC integrado, el receptor de pantalla ancha y el receptor de HDTV (alta definición). La parte de datos digitales de la señal ofrece una diversidad de opciones de sonido de alta calidad así como servicios de datos que representan un área de creciente interés. Ofrece más oportunidades para aportar ganancias para el radiodifusor y en el contexto de HDTV ofrece un canal de control por el que puede efectuarse una mejor decodificación de recepción.

Comenzando hacia atrás desde los receptores y el sistema de transmisión, se requiere estudios y fuentes para alimentar los servicios nuevos y existentes. En el entorno del R.U. el estudio estándar ideal de HDTV utilizará 1250 líneas y una frecuencia de repetición de campo de 50Hz. Esto se interconectará bien con el entorno de vídeo y film existente y es idealmente adecuado como fuente para un sistema de condificación MAC de alta definición para la transmisión de HDTV.

Revision general del sistema de Paquete D-MAC

Resumen

En este capítulo se da una valuación del concepto global del estándar de transmisión D-MAC. Se considera la naturaleza de los Componentes Analógicos Multiplexados siguiendo con la descripción de la forma de onda de visión, los motivos para la compresión de tiempo y los regímenes de muestreo elegidos. Antes se dan los detalles de los canales de sonido y datos, prestándose también atención a la elección de la condificación duobinaria y la formación de la estructura de paquetes. Se menciona la base de las técnicas de mezcla y encripción siguido por los requisitos de cable para D y D2-MAC.

Creemos que después de leer este capítulo el lector podrá apreciar el contenido de cualquiera de los otros capítulos sin necesidad de considerarlos en un orden determinado.

La señal de vision

Resumen

Dentro de este capítulo se hace una comparación entre la estructura de D-MAC y PAL. Se describe el formato de la forma de onda de banda base D-MAC junto con la composición de la trama completa de 625 líneas. Se dan razones para el uso de compresión de tiempo de visión y se discuten las consecuencias del ancho de banda así como de la transmisión secuencial de la señal de diferencia de color. Este capítulo incluye también secciones dedicadas a sincronización, fijación, perturbación de visión y dispositivos para la transmisión de imágenes de más amplia proporción de aspecto.

Construccion y transmision de paquetes

Resumen

Se pasa revista a las bases de los posibles formatos de codificación de sonido/datos antes de la adopción del formato de paquete D-MAC. Antes de la estructura de paquete D-MAC el capítulo considera las técnicas básicas de la codificación duobinaria y cómo puede conseguirse un aumento en la capacidad de bit. Se describe la estructura de un paquete individual incluyendo las funciones de la cabecera de paquetes junto con la necesidad ocasional del tipo de paquete de 8 bit. Se considera la naturaleza flexible del múltiplex con ilustraciones para ver cómo una trama completa de dos subtramas de sonido/datos pueden soportar 4100 paquetes por segundo dando una velocidad de bit media de 2,952 Mbits/s. Se discute también los usos alternativos de dos subtramas junto con los requisitos de la transmisión de paquete.

Canales de sonido

Resumen

La capacidad del canal de sonido/datos de D-MAC permite hasta ocho canales de sonido de alta calidad o dieciséis canales de comentario, todos los cuales pueden ser acompañados por datos con capacidad en exceso de un canal terrestre existente. Cada canal de alta calidad puede tener uno o dos niveles de protección de error y puede ser codificado linealmente o en NICAM. Este capítulo da una descripción de estas posibilidades con un tratamiento simplificado de la opción de codificación de NICAM.

Se describe la inserción de varios canales de sonido en la estructura de paquete junto con requisitos globales de múltiplex de sonido/datos.

Bloques de interpretacion (BI)

Resumen

La información llevada en los Bloques de Interpretación permite que el decodificador de sonido proporcione automáticamente la salida correcta de audio para el servicio seleccionado y prepare el decodificador para cualquier cambio próximo en el servicio seleccionado. Esta información, en forma de grupos de datos, es conducida por paquetes en el múltiplex de sonido/datos. Se describe la composición de órdenes y su organización en grupos de datos, junto con un ejemplo del funcionamiento del Bloque de Interpretación.

Señales de interpretacion de servicio

Resumen

La información sobre el canal de transmisión por satélite junto con detalles de los servicios de visión, sonido y datos disponibles son llevados al receptor por medio del canal de Identificación de Servicio. Esta considerable cantidad de información es ensamblada en filas desde el parámetro básico de una orden, hasta el formato complejo de un Grupo de Datos. Lo mismo que los otros datos en el múltiplex de sonido/datos esta información es llevada por paquetes. Este capítulo sigue la ruta de un parámetro a través de las capas de la estructura mientras procura una descripción de cada etapa. Al final del capítulo se da un ejemplo de un canal de satélite típico y la capacidad requerida para describir sus servicios.

Radiodifusion de datos

Resumen

La ráfaga de sonido/datos del sistema de paquete D-MAC puede ser configurada para llevar no sólo sonido y los datos necesarios para manejar el múltiplex global sino

también para transportar teletexto, subtítulos, imágenes fijas y canales de datos transparentes y mucho más. El sistema es muy flexible y puede hacerse todavía más utilizando el campo completo para transmisión de datos cuando no se requiere la señal de visión.

El contenido de este capítulo está dedicado principalmente a la construcción de dos formas de bloque de datos de teletexto. Ambas están derivadas de la Recomendación de CCIR 653 sistema B (Sistema de Telexto Mundial) pero difieren en la manera de sus niveles de protección. Se da una descripción de los niveles de protección y cómo se inserta un paquete de teletexto completo en un paquete de D-MAC junto con la señalización dentro del canal de Identificación de Servicio.

Informacion de alta prioridad – Linea 625

Resumen

La función principal de esta línea, que consiste completamente de datos, es organizar la estructura de toda la transmisión múltiplex. Los datos dentro de la línea son de dos categorías principales. Primeramente, información estática, la cual si bien es importante para el receptor, es básicamente constante de trama a trama. En segundo lugar, información dinámica que cambia y debe por tanto ser llevada rápidamente al receptor trama a trama. Ejemplos de información llevada por el bloque de datos estáticos son la identificación de la posición del satélite junto con la fecha y la hora. La información dinámica describe la configuración del múltiplex que puede cambiar de cuando en cuando dependiendo de los requisitos de los radiodifusores.

Se dispone de una lista de los diversos bloques de datos dentro de las dos categorías junto con una descripción de sus funciones.

El sistema de acceso condicional

Resumen

Las transmisiones por satélite pueden hacerse ininteligibles para el usuario por un proceso conocido comúnmente como perturbación. Sin embargo, la perturbación solamente es insuficiente si se desea un formato de pago por vista para el operador de canal. Las mismas señales pueden ser perturbadas, pero se necesita algún dispositivo para enganchar y desenganchar el proceso de perturbación, lo que se conoce como encripción. El capítulo describe ambos de estos procesos y la manera en que las 'claves' de desenganche pueden pasarse a través de canales de transmisión separados al receptor. Se discute un sistema de pago por vista seguido por una implantación práctica de un sistema de direccionamiento sobre aire de clave compartida junto con una sugestión de cómo eliminar el pirateo de programas. Se considera también los requisitos de transmisión y de capacidad de datos.

Transmision de RF de Paquete D-MAC

Resumen

Este capítulo describe el rendimiento de una señal de paquete D-MAC en un canal de satélite. Esto se ilustra en forma tabular con la adición de gráficos mostrando los resultados subjectivos de calidad de imagen bajo relaciones de portadora a ruido variables. Se hace también una comparación entre el rendimiento de senales de sonido en un canal de 27 MHz y en un canal de 21 MHz. En el primer caso se utilizan codificaciones de paridad de Hamming y NICAM incorporando las técnicas de decodificación convencionales, mientras en el último caso se muestra que la decodificación de sonido de Viterbi permite mejorar la recepción de relaciones de portadora a ruido más bajas.

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* Out of Print

THE INDEPENDENT BROADCASTING SYSTEM

The Independent Broadcasting Authority (IBA) is the central body responsible for the provision of Independent Television (ITV, including TV-am, and Channel 4) and Independent Local Radio (ILR) services in the United Kingdom. The IBA selects and appoints the programme companies; supervises the programme planning; controls the advertising; and builds, owns and operates the transmitters. Independent Broadcasting is completely self-supporting, financed by the sale of spot advertising time in the companies' own areas. By late 1989, British Satellite Broadcasting (BSB), will launch new programme services on three channels broadcast directly from a satellite (DBS).

More information is contained in the IBA's Annual Report 1987-88.

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